THE DYNAMICAL ANALYSIS OF SPEECH SOUNDS

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PREFACE

A steady tone at a given point in the sound field may be represented in two equivalent ways, either by the instantaneous pressure as a function of time or as the sum of harmonically related pure tones, i.e., tones having a sinusoidal wave form. The latter form of representation is called the spectrum of the time function. The process of transforming a periodic time function into its spectrum is known as harmonic analysis. A problem in acoustics originally suggested this now widely applied type of transformation. As a result of his study of the vibration of a string, a basic element in the earliest forms of musical instruments, Daniel Bernoulli enunciated the principle of the co-existence of small oscillations and declared his conviction that the general solution of the free vibration of a string could be expressed in the form of a sine series. Euler a little later supplied a method for evaluating the coefficients in the series.

One reason why the harmonic analysis plays such an important role in acoustics is that the ear, according to Ohm's law, perceives a tone as a summation of harmonic components, in other words, it performs an harmonic analysis of a periodic acoustic stimulus.

The harmonic analysis can be applied in a simple, straightforward manner to a steady time function, such as a tone. It is thereby transformed completely into a frequency function. Less direct is its application to more complicated functions such as speech or music.

Fourier in his celebrated work on the application of the trigonometric series to physical problems demonstrated that a series of this kind can always be fitted exactly to any small or large finite portion of a single-valued function. Therefore if a function is constituted of a succession of steady, or tonal, portions together with some portions that are random in character, or atonal, a condition that obtains for speech and music, then the Fourier theorem may be applied to this function either as a whole or successively to small portions of it. The first procedure leads to a single sine series and the second, to a summation of sine series each of which by itself is valid only over one particular sub-interval of time. In this latter case the transformation is not completely from a time function to a frequency function but to a time pattern of frequency functions. I have designated this type of transformation a dynamical analysis because of the time aspect of the result.

From our everyday experience we can infer that it is a dynamical type of analysis that the ear performs on speech, for it follows rapid changes in pitch, quality and loudness. Since the ear can with great success extract the information content from speech, I have in the work described on the following pages attempted to develop quantitatively the properties of this analyzing process of the ear on the basis of available hearing data. This study is followed by a description of apparatus which was designed to have analyzing properties like the ear but with the results expressed in quantitative physical terms. Finally, the results of analyses made by this equipment on the principal sounds of speech under a variety of phonetic conditions are given.

The detailed planning and execution of this work extended over a period of several years. During that time I received much valuable advice and assistance from more people in various departments of the Laboratories than I could possibly recount. But I cannot fail to acknowledge my indebtedness to Dr. Harvey Fletcher, retired Physical Research Director, to whom I owe most of my interest in the physics of speech, and to Dr. Harold T. Friis, Director of Research, High Frequency and Electronics. Except for his stimulating personal interest this work would not have been completed as planned.

I gladly acknowledge the valuable aid received from Dr. H. K. Dunn in the analytical study of the moving coil drive for a level recorder, from Mr. J. R. Nelson in the design and testing of most of the electrical circuits used in the equipment, and from Mr. A. H. Müller in the design of the mechanical parts of the apparatus and for his supervision of their construction. More than to any other single individual who worked with me on this project I am indebted to Mr. J. H. Kronmeyer, who was associated with me in this work from its beginning. He assembled the apparatus and did most of the trouble-shooting and calibrating. He also made all the recordings and did most of the work connected with the assembly of the data as it has been presented.

To Professor G. E. Peterson of the University of Michigan, Professor and Mrs. Malcolm Coxe of Brooklyn College I am indebted for helpful discussions regarding the selection of material for analysis, its presentation, and other phonetical aspects of the problem.

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> "It is reason and speech that unite men to one another; there is nothing else in which we differ so entirely from the brute creation." - Cicero.

Speech is such an important factor in the civilization of man that some writers have held with good reason that speech is civilization. It has therefore been studied with the aim of increasing its effectiveness in one respect or another throughout all historical times. In the third millenium B.C. The Instructions of Ptah-hotep was written in Egypt for the purpose of "instructing the ignorant --- about the rules of good speech, as of advantage to him who will hearken and of disadvantage to him who may neglect them."[1] Ancient Greek scholars placed great emphasis on the study of rhetoric, the skillful and artistic use of speech. One application of this interest was in the drama, an early form of art. Open-air theatres that were built by the Greeks and Romans, some of them long before the Christian era, showed in their design an appreciation of certain acoustic principles that must be followed if speech delivered from the stage is to be readily understood by a large audience.[2] Vitruvius, who lived at the time of Augustus, discussed clearly some of these principles in his celebrated work on architecture. A development in quite another direction for making speech more effective was the invention of the megaphone. Alexander is said to have used megaphones for the transmission of speech over distances as great as 12 miles.

While civilized peoples have always been interested in ways of improving the transmission of intelligence by the spoken word, the objective study of speech itself began only within the last century. The first physical measurements on speech were made with the phonautograph invented by E. Scott.[3] In this instrument a diaphragm was actuated by speech waves collected through a horn. The resulting motion of the diaphragm was recorded by an attached stylus resting on a moving strip of soot-coated paper. Judged by present standards the distortions in this instrument must have been large and the record therefore, a crude approximation to a facsimile of the initial speech waves. Nevertheless certain kinds of quantitative values, such as frequency and duration, could be read from the records.

Basic Principles of Speech Analysis

It is a remarkable fact that we are able to understand anyone who speaks clearly in a familiar language although the speech of any individual person differs unmistakably from that of everyone else. We therefore naturally ask: "What are the characteristic physical properties of any given standard speech sound whereby the ear recognizes its meaning?", or, in the language of the phonetician, "What physical properties that the ear can sense characterize any particular phoneme?" This is not only an intriguing question of particular interest to the phonotician and the physicist, but also one of practical importance to the telephone engineer and to anyone interested in the design of voice-operated devices.

Since the ear is so remarkably effective in the identification of the speech sounds, we should do well to inquire into the type of analysis which the ear performs in isolating the distinguishing physical characteristics of these complex stimuli. Having obtained this information about the ear, we should then be able to devise instrumental means for making the same kind of analyses in quantitative physical terms.

Before we consider the analysis of the very complicated waves of speech, let us first review what is known about a much simpler problem, viz., the aural analysis of complex but steady periodic tones. It had long been known to mathematicians, even before Fourier's time, that a continuous periodic function can be expressed as a series of sine functions, but it was not until 1843 that Ohm announced his now well known law of hearing, which states that the ear perceives as pure only those tones which have a sinusoidal wave form while all others are heard as the summation of pure tones. This law was readily verified for the lower order of harmonics. By a careful focusing of attention, one could rather easily hear the octave, the fifth, and second octave above, i.e, the second, third, and fourth harmonics, in the tones of some musical instruments; but the general validity of the law was not accepted until the publication of Helmholtz's experimental studies of the subject. Helmholtz apparently became particularly interested in the problem because Seebeck, a distinguished acoustical physicist, had come to believe that while the ear perceived separately certain harmonics in a complex tone, there were some groupings of harmonics which were perceived as a unit. This view was not unreasonable because, when several musicians together play a melody on separate instruments, a listener by properly focusing his attention can distinguish the tones that come from each of the instruments. In this case it appears that the ear is resolving the complex tones coming from the ensemble into other not pure but complex tones. Helmholtz gave a correct, if not complete, explanation of this observation and then pointed to Leibnitz's distinction between apperception and perception, which holds for all the senses. As applied to aural sensation, it means that a listener ordinarily senses any complex tone not as a composite but as a unitary sound having a distinctive quality. This type of observation Leibnitz called perception. As an observer becomes aware of the fact that he can hear harmonic components within the complex tone, he gradually learns how to focus his attention on them and one by one is enabled to hear them individually. This manner of observation Leibnitz called apperception. When one proceeds from perceiving to apperceiving, the focusing of attention is the important factor, as Helmholtz recognized. He therefore devised an acoustic resonator which is now known by his name. With this resonator he could raise the relative level at the ear of a particular component in the tone. Holding his attention on the component thus made prominent, he could remove the resonator and still hear the component separately from the rest of the tone. By this procedure he was enabled to resolve musical tones up to the 16th harmonic. By modern techniques it is possible to direct attention to and then hear components of much higher order within complex tones that are rich in harmonics. There is no longer any doubt about the validity of Ohm's law.- See, however, reference 33.

Ohm's law implies that the relative phases of the harmonic components of a complex tone have no effect on its tonal quality. Helmholtz also investigated this question with musical tones and found no phase effects.* The ear in effect thus transforms the incoming complex tone into a line spectrum, rejecting whatever information might be carried by the phase relationship between the components.

Ordinarily we think of a line spectrum as a chart on which the Fourier components of a periodic function are represented by the length of vertical lines placed in sequential order on the axis of obscissae at equally spaced intervals. Except for phase such a chart is an accurate pictorial representation of a periodic function. Since all the harmonic components occupy an equally prominent position in such a chart, it also gives a well-balanced picture of any periodic function in which all the components are of equal importance. But such is not the case with regard to the components of a complex tone as perceived and evaluated by the ear.

While Ohm's law states that a tone is perceived as the sum of its harmonic components, it tells nothing about the relative amount that each contributes to the production of a particular tonal quality. Obviously any components lying outside of

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* For interesting exceptions to this rule under a number of special conditions the reader should consult a paper by Mathes and Miller, J. Acoust. Soc. Am. 19, 778 (1947).

audible range could contribute nothing, but how is the importance of the components lying within this range weighted by the ear? A casual observation reveals that the weighting is not uniform. Consider, for example, a 100 cps tone that is rich in harmonics up to several thousand cps. It would be found that the elimination of a component near the top of the range would alter the quality of the tone very little in comparison with the change that would be produced by elimination of the second or third component. Likewise the difference between two tones in one of which the second and in the other of which the third harmonic is missing is much greater than the difference between two tones in one of which the 29th and in the other of which the 30th component is absent. In other words a single low frequency component has in this case as much effect on tone quality as a whole group of high frequency components. This fact suggests that a graphical representation of a steady complex tone most representative of what the ear perceives is one in which the audible frequency range is divided into bands that are of equal importance in their effect on tone quality and in which the mean total power in each of these bands is plotted against band order number. In addition to the power distribution the chart should in some way show the fundamental frequency of the tone. This requirement will be discussed further in a later paragraph. Since sound is perceived by the ear approximately in accordance with Fechner's law, the power values should be shown in decibel or other logarithmic units.

From the above discussion and from our general feeling that an octave represents the same pitch interval at all parts of the musical scale, we might easily conclude that bands of equal importance for quality would cover equal pitch intervals; but all pertinent experimental data on hearing suggest a somewhat different frequency division. A relatively simple scale, expressible in objective terms, that fits the subjective characteristics of the ear remarkably well over the frequency range of importance to speech intelligibility, i.e., 200-7000 cps, is one proposed by W. Koenig.[10] This scale is linear in frequency below and logarithmic above 1000 cps. Over the stated frequency range it has the property that equal intervals anywhere along the scale represent an equal number of just perceptible steps in pitch, an equal number of mels*, equal fractions of the critical bandwidths+, and bands of equal importance in respect to the intelligibility of speech. There are also some direct experimental data indicating

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* The mel scale is a subjective scale of pitch based on the judgment of fractional values of pitch intervals. - Stevens, Egan, and Miller, J. Acoust. Soc. Am. 19, 775 (1947).

\+ A critical band is defined as the minimum band of white noise that will produce as much masking of a pure tone of mid-band frequency as a broad band of the same noise. - H. Fletcher, Speech and Hearing in Communication, D. Van Nostrand, Inc. (1953) p.171.

that bands of frequency on the Koenig scale represent bands contributing equally to the perceived quality of a tone.[11] Judgment tests were made with a 100 element tone synthesizer of the effect on tone quality produced by elimination of groups of harmonics in various parts of the frequency spectrum. The synthesizer was set to develop harmonics of 100 cps. While these tests were not so extensive as would have been desirable, the data obtained with three or four observers indicated that harmonics grouped within equal intervals of the Koenig scale contributed equally to the quality of the tone. In the following discussion we shall therefore consider power level (db) values in equal bands, i.e., in equal intervals, of this scale.

Before modern electro-acoustic instruments became available, much effort was spent in making Fourier analyses of typical cycles in vowel records by mechanical or graphical methods. Such studies have also been made of oscillographic records produced by modern high quality instruments. This procedure becomes very tedious when the number of harmonics is as large as it is in the vowel sounds. This difficulty was overcome in the electro-optical method of analysis developed by C. F. Sacia[12]. In this method, as applied by Crandall and Sacia[7] to the study of vowel sounds, records were made of the vowels on strips of photographic film with a high quality microphone, amplifier, and oscillograph. The record of a particular vowel so recorded was reproduced electrically in quick repetition by specially designed equipment. The generated current was studied by means of tuned circuits. From these measurements an amplitude-frequency spectrum was obtained of the complete vowel as spoken instead of that of a typical cycle. Since the publication of the work of Crandall and Sacia a number of other types of current analyzers have been developed for use in the study of acoustic problems. In most of these instruments the selective circuits have been designed to give band-pass filter rather than resonant circuit characteristics. Some of them are of fixed band-width, of which those operating on the heterodyne principle are typical. In other analyzers the band-width is varied with the mid-frequency of the band. For example, of the set of 13 band-pass filters in the analyzing equipment used by Sivian[13] for the statistical study of speech power, those transmitting below 500 cps had a band-width of one octave while the rest had a band-width of a half octave. More recently Stevens, Egan, and Miller[14] also used, in effect, 13 band-pass filters for the study of speech. Their filters were designed to effect a division of the speech power into frequency bands of equal width on the mel scale. This scale, as we have noted, closely approximates the Koenig scale through the speech frequency range.

Thus far we have considered only the analysis of a steady tone, but the best form of representation of the way the ear perceives a steady noise also is a chart showing the power level in bands of equal width on the Koenig scale. This conclusion is based on the results of studies on the loudness and masking of this type of noise; pertinent particularly is the fact that the critical bands are of equal width on this scale.

We have devoted several pages to discussing the visual representation of steady state sounds, but speech sounds are mainly transient in character. Only for short intervals of time are there any steady portions. However, according to Bürck, Kotowski and Lichte[15], experimental data on short pulses indicate that there is no fundamental difference between the ways in which the ear analyzes a tone and a transient. For a tone the analysis gives a Fourier series and for the transient a Fourier integral. We can therefore assume that the same type of analyzer can be used for obtaining aural type of information on both steady and transient sounds.

The schematic form that such an analyzer might take is shown in Fig. 1. This is in principle the same as that of the apparatus used by Crandall and Sacia[7] for their vowel studies. A microphone is connected to the input of each of n filters which together cover the speech frequency range. The output of each filter is led to a separate level recorder. Means have to be provided for keeping the recording papers of the bank of recorders moving in synchronism so that corresponding instants of time will be identifiable on all the records.

While it is true that the type of analyzer shown schematically in Fig. 1 may be used for the analysis of speech as well as for steady sounds, the filter requirements are generally different. The steady state filters should preferably have sharp cutoff characteristics. As we shall see, transient oscillations in filters of this type are relatively large and decay slowly so that they can mask important details of the filtered speech currents. This possibility must be avoided in the design of the speech filters.

It is well known that the rate at which the output current of a filter can be varied is inversely proportional to its transmission band-width. The choice of band-widths in the design of the speech filters must therefore involve a compromise between the maximum time rate with which the output current of the filters can follow changes in input level and the precision with which the distribution of power over the frequency range during the relatively steady parts of speech can be determined. This compromise must be based on both the physics of speech and the sensory characteristics of the ear.

In the following discussion of electrical networks and equipment we shall find it convenient to use the terms frequency-response and time-response. The former we define as the difference in level between the output and the input power expressed as a function of frequency, and the latter as the transient output current that follows a change of unit input voltage.

A general treatment of transient oscillations in wave filters is given in a highly technical paper by Carson and Zobel[16], to which the reader interested in the fundamental aspects of these phenomena is directed. Their conclusions are applicable to all types of selective circuits, although they treat particularly the case where the wave filter is terminated in an impedance equal to its own characteristic impedance. Under these conditions it has a sharply defined transmission band, but, as they remark, "its selective properties are essentially properties of the steady state only".

We can get a broad concept of the general nature of transient currents in filters from actual records of such currents in practical filters of known frequency-response. Fig. 2 shows the frequency-response of several band pass circuits. These are all plotted so that zero on the scale of abscissae corresponds to the mid-band frequency, and one to the frequency at which the response is down 3 db from its value at mid-band. This latter frequency will be called the cutoff frequency and will be symbolized by ω_c for all filter type networks considered in the following discussion. The curve marked "a" shows the transmission of a well designed filter having low attentuation inside and high attenuation outside the nominal transmission band. This band is the half octave from 500 to 700 cps. Fig. 3a shows the time-response for this filter when the input power at mid-band frequency is abruptly lowered by 60 db. This record was made with a high speed level recorder, which will be described in a later section, while the filter was terminated in a resistance numerically equal to its characteristic impedance. The dotted curve shows the rate at which the current level in a simple series circuit, tuned to a frequency of 600 cps and having a band-width of 200 cps, would decay. The curve "b" of Fig. 2 shows the transmission characteristic of a heterodyne sound analyzer having a crystal filter of 75 cps band-width. The decay curve of the output level when the input voltage was suddenly reduced by 60 db is shown in Fig. 3b. The voltage in this test had a frequency of 1000 cps, to which the mid-band of the analyzer was set. The dotted curve gives the rate at which the current level would decay in a simple series tuned circuit having a resonance frequency of 1000 cps and a band-width of 75 cps. The frequency response of such a resonant circuit is shown by R in Fig. 2. The foregoing response curves show that a simple resonant circuit is relatively fast acting but has low selectivity, whereas the classic filters have frequency

responses approaching the "ideal" but exhibit transient currents of relatively long duration. Neither type of selective network appears to be satisfactory for use in the dynamical analysis of speech sounds. The problem of finding a type of filter which will have a fast time-response with the least sacrifice in sharpness of cutoff will be discussed in the next section. For the moment we shall assume that filters of optimum properties are available and consider some of the other requirements that the analyzing apparatus must satisfy.

Since the ear response is independent of the phase relation between the components of a tone, the readings of the recorders connected to the output circuits of the filters should also indicate the value of a function of the components that is independent of their phase. The only function that satisfies this requirement is one that depends upon the mean value of the power. The term mean as used here requires some clarification. If the mean power readings were to show no variation with phase whatever, the power would have to be averaged over an infinite time period or at least over the duration of the signal; but then the readings would not show the significant temporal variations in power. On the other hand, if the power were averaged over a very short time interval, the readings would indicate practically the instantaneous values of power, which would not be independent of phase. We can put the matter in another way by saying that the recorder on each channel should record the envelope of the power wave, but we need to specify how closely it should follow the finer details of this envelope. Here there are two determining factors. The record must show the power variations in sufficient detail to indicate qualitatively whether the power is predominantly periodic or random, i.e., whether it corresponds primarily to a tone or a noise. If we want to be sure that the capabilities of the filter can be fully exploited, the recorder must be able to follow closely its time-response.

Previous studies on speech intelligibility have shown that the components of speech sounds which contribute to its intelligibility may have power values that extend through a range of 50 or 60 db. Because of this wide range of power levels which the recorder must comprehend and the fact that the sensory perception of the ear obeys the Weber-Fechner law approximately, the recorder should operate on a logarithmic scale, preferably divided into conventional decibel units.

From previous studies we also know that speech waves are unsymmetrical and that the peak power in a period of these waves may be 10 to 15 db above the mean power for the period.[17] The recorders must be designed so that they will indicate correctly the mean values of the power in waves of this character.

Filter Design

An "ideal" filter may be defined as a four-pole network which has no attenuation within a certain frequency band and infinite attenuation outside this band. The flat frequency response in the transmission band implies that within this interval the time delay in the filter is independent of frequency and a signal would be transmitted without distortion of wave-form. However, the "ideal" filter cannot be realized exactly, not even theoretically. There will always be a transition interval between the pass and no pass regions in which there is considerable phase distortion.[18] This phase distortion in the cutoff region can cause delayed transient currents of considerable magnitude, such as those shown in the time-response curves of Fig. 3, whenever any part of the power in the signal falls within this frequency interval. Therefore in the design of band-pass filters that are to be used for subdividing power covering a wide frequency range in a random manner, such as that of speech currents, a balance must be sought between sharpness in frequency discrimination and freedom from delayed transient currents.

An analyzer of the type shown in Fig. 1 comprises a set of band-pass filters; but in trying to determine the optimum attainable properties of one of these filters, we shall first consider the simpler low-pass filter. Afterwards, band-pass filters of equivalent characteristics and of desired mid-band frequencies will be derived by simple transformations.

The problem of designing this low-pass filter so that it may have the best possible characteristics is quite similar to that of designing a video amplifier. Amplification is of course not a requirement in the filter design, whereas it is the prime function of the video amplifier; also the frequency ranges covered are far different, but in both cases the following properties are desired: a flat frequency response and the fastest possible time-response with limited overshoot in the operating frequency band, and negligible response outside this band. In the design of the filter we can therefore benefit by recent studies of video amplifiers. Of particular interest is a paper by Kallmann, Spencer, and Singer.[19] These investigators studied the frequency and time-responses, and the delay time characteristics of many types of both physically available and hypothetical interstage coupling networks. From their data one can at least surmise that a filter having a frequency-response approximately that of a Gaussian probability function would incorporate the best possible compromise of the desired properties as stated above.

The frequency-response of a Gaussian low pass filter is given by the expression

$$A = A_o e^{-\left(\frac{\omega}{\omega_e}\right)^2}$$

where A_o is the dc response and ω_e is the frequency at which $A = A_o/e$, a disposable constant. The cutoff frequency, ω_c, of this filter is equal to 0.589 ω_e. The frequency-response is shown graphically as a function of ω/ω_c by G in Fig. 2.

It can be shown[19] that the time-response of this filter is

$$E = \frac{1}{2}\left[1 + \varphi\left(\frac{\omega_e t}{2}\right)\right] A_o$$

where $\varphi(x)$ is the error integral, an odd function, for which the values may be found in books of mathematical tables. The time-response as computed by this formula is shown by the dashed curves of Fig. 4 and Fig. 5, respectively in linear and in logarithmic (db) units. The transition curve in linear units is seen to be symmetrical with no oscillations or overshoot.

We shall appropriately at this point refer to some of A. Gabor's work on communication theory[20]. R. V. L. Hartley[21] in 1928 had already shown that the measure of the capacity of a system for transmitting information is the product of the width of its frequency range and the time that it is available. Gabor, interested in obtaining a quantative definition of the elementary datum of information in a signal, showed that, if Δf is defined as the root mean square frequency-width and Δt as the root mean square duration of the power of a pulse, then there is a definite minimum value below which the product $\Delta f \times \Delta t$ cannot be brought. He furthermore showed that the product has this minimum value if the pulse has the form of a Gaussian probability curve. This form of pulse has a number of other properties which are of special interest in the communications art[22]; one of these is the self reciprocal character of its Fourier transform[23]. Its time and its frequency patterns are therefore both of the probability form. For this reason and because a unit infinitesmal pulse has a uniform spectrum, it follows that, if such a pulse is applied to the input of a Gaussian filter, the resulting output will be a pulse of probability form. By the principle of superposition, which holds for a linear system, any signal voltage applied to the input of the filter can be expressed as the integral sum of corresponding probability pulses, having the minimal properties pointed out by Gabor.

While Gabor's definition of pulse duration and Kallmann, Spencer, and Singer's definition of transition time may not be most representative of the way these properties are perceived by the ear, we can nevertheless be reasonably sure from the analytical studies of these investigators that a filter of optimum characteristics for speech analysis will have a frequency-response that is not far from Gaussian in form.

If a unit pulse were applied to a Gaussian filter, the resultant output pulse would stretch from $-\infty$ to $+\infty$ in time. This fact shows that such a filter could be realized only in a circuit having an infinite number of elements or in an infinitely long line. The range of both levels and frequencies that are of interest in speech analysis are limited. It is possible to obtain Gaussian characteristics over these restricted ranges to any desired degree of approximation with a finite number of circuit elements. One procedure for accomplishing this end is indicated by C. E. Cherry[24], who suggests the use of a cascade amplifier with resistance-capacitance interstage couplings and shows that, if the anode resistance is high compared with the network impedance, the frequency-response of the amplifier will approach that of a Gaussian filter as the number of stages is increased. For a good fit over the relatively large level range of speech, the required number of stages would be rather large. Dr. H. W. Bode pointed out to us that the same result may be obtained with any type of interstage coupling so long as it has a monotonically drooping frequency-response and that therefore a given approximation to a Gaussian characteristic may be attained in fewer number of stages by the use of more potent coupling networks. After a further helpful discussion of this problem with Dr. R. L. Dietzold, we came to the conclusion that an amplifier with similar series peaking interstage networks afforded the most practicable way of obtaining filters of desired Gaussian properties. The coupling of 2 triodes by this circuit is shown in Fig. 4. It is assumed that the anode resistance is large compared with the resistance, R, of the network. If a unit voltage is applied to the grid of the first tube then the output voltage of the first coupling network will be

$$V_1 = \left[1 + \left(\frac{1}{Q^2} - 2\right)\left(\frac{\omega}{\omega_o}\right)^2 + \left(\frac{\omega}{\omega_o}\right)^4\right]^{-\frac{1}{2}} V_o$$

where $\omega_o = \sqrt{\frac{1}{LC}}$, $Q = \frac{\omega_o L}{R} = \frac{1}{\omega_o CR}$ and V_o is the output voltage for dc. If there are n equal stages, then the output voltage of the nth network will be

$$V_n = \left[1 + \left(\frac{1}{Q^2} - 2\right)\left(\frac{\omega}{\omega_o}\right)^2 + \left(\frac{\omega}{\omega_o}\right)^4\right]^{-\frac{n}{2}} V_o^n \quad - - - - \qquad (1)$$

From the solution of the differential equation of the circuit we find that, when a unit dc voltage is suddenly applied to the grid of the first tube, the transient output voltage of the first network is

$$E_1 = \left[1 - \frac{\omega_o}{\omega_1} e^{-\Delta t} \cos(\omega_1 t - \varphi)\right]$$

in which $\Delta = \frac{R}{2L} = \frac{1}{2}\frac{\omega_o}{Q}$

$$\omega_1^2 = \omega_o^2 - \Delta^2$$

$$\varphi = \sin^{-1}\frac{1}{2Q} = \tan^{-1}\frac{\Delta}{\omega_1}$$

If this one stage is coupled to a second stage with a similar network, then the time-response of the combined stages may be found by application of the principle of superposition through the Duhamel integral[25] as follows:

$$E_2 = \int_o^t E_1(t-\lambda)\, E_1'(\lambda)\, d\lambda$$

$$= 1 - e^{-\Delta t} \frac{\cos(\omega_1 t - \varphi_1)}{\cos\varphi_1} + \frac{1}{2} e^{-\Delta t} \left(\frac{\omega_o}{\omega_1}\right)^2 \omega_o t \sin(\omega_1 t - \varphi)$$

in which $\varphi_1 = \tan^{-1}(6Q^2-1)/(4Q^2-1)^{3/2}$

If two more stages are added, i.e., if the number of stages is doubled, with similar coupling networks, we can again apply the Duhamel integral and get the time-response of the four combined stages:

$$E_4 = \int_o^t E_2(t-\lambda)\, E_2'(\lambda)\, d\lambda$$

Substituting E_2 we get:

$$E_4 = 1 - M \frac{2Q^4}{(4Q^2-1)^3} e^{-\Delta t} \qquad (2)$$

in which

$$M = \left[\frac{(4Q^2-1)^3}{2Q^4} - \frac{38Q^4-11Q^2+1}{Q^2\sqrt{4Q^2-1}}\omega_1 t - (6Q^2-1)\omega_1^2 t^2 + \frac{2}{3}\frac{Q^2}{\sqrt{4Q^2-1}}\omega_1^3 t^3\right]\cos\omega_1 t +$$

$$\left[\frac{140Q^6-70Q^4+14Q^2-1}{2Q^4\sqrt{4Q^2-1}} + \frac{22Q^4-9Q^2+1}{Q^2}\omega_1 t - \frac{8Q^2-1}{\sqrt{4Q^2-1}}\omega_1^2 t^2 - \frac{2}{3}Q^2\omega_1^3 t^3\right]\sin\omega_1 t$$

An expression for the time response of an 8 stage amplifier could be obtained similarly by substitution of E_4 in the Duhamel integral, but it would lead to a very complicated algebraic expression which would not serve a useful purpose, as this number of stages is impracticably large. The derivation of an explicit expression of the time-response for an intermediate number of stages is difficult if not impossible. We shall therefore study the responses of the four stage filter and from the data so obtained decide upon the number of stages that should be incorporated in the speech filter-amplifiers.

The frequency-response curves for a four-stage filter-amplifier as computed by (1) are plotted in Fig. 6 for 3 different values of Q. ω_o was in each case set so that the cutoff frequency, ω_c, had the same value. For comparison purposes the frequency-response of a Gaussian filter is indicated by a dashed line. Of the three filters represented in this figure the response of the one having coupling networks with a Q of 0.56 comes nearest to this line, particularly in the more important region of low attenuations. The frequency-response of a five stage filter having coupling networks with a Q of 0.56 is also shown and is seen to follow the dashed curve still more closely. Fig. 4 gives the time-responses of these four section filters as computed by (2). The curve corresponding to a network Q of 0.6 has a noticeable but small overshoot; otherwise all three curves appear to be good approximations to the time-response curve of a Gaussian filter, which is shown by the dashed line of the figure. The highest possible value of Q for absolutely no overshoot is 0.5. Fig. 5 shows the same data plotted with the ordinates on a logarithmic scale.

Regarding the filters to be used for speech analysis, we are not so much concerned about the output current resulting from an increase in voltage from zero to some finite value as we are in knowing its form when the input level is changed from one value to another over a maximum range of the order of 50 or 60 db. A sudden increase in the dc input voltage of a four stage filter by a factor, m, would produce an output emf,

$$E_m = \frac{1}{m-1} + E_4(t) \text{ - - - - - - } \qquad (3)$$

The curves of Fig. 7, which were plotted from (3), show the build-up and decay output voltages resulting from sudden changes in the input voltage level of a four stage filter-amplifier having coupling circuits with a Q of 0.6. There is no appreciable overshoot in the build-up curves but a rather large, although brief and well-damped, overshoot in the decay curve for a change in input voltage of 40 db or more. It may at first seem paradoxical that there should be more overshoot in the decay than in the build-up curve since we have assumed a linear system in which the principle of superposition is valid. If the curves had been

plotted on a linear ordinate scale, the observed overshoot would have been, in fact, the same in the two cases. The difference becomes so conspicuous in Fig. 7 only because the output voltage is plotted on a logarithmic scale. Fig. 8 shows the build-up and decay curves for a four stage filter in which the coupling networks have a Q of 0.56. The overshoot is seen to amount to only a few db even when the drop in input voltage level is as much as 50 db. The corresponding curves for a Gaussian filter are indicated by the dashed lines, the two sets of curves having been arbitrarily brought into coincidence at the 44 db level.

From the theoretical data here presented it appears that a four stage filter amplifier with series-peaking coupling networks will have most nearly the optimum characteristics required for speech analysis if the Q's of the networks have a value of about 0.56. We therefore decided to use filter amplifiers having coupling networks of this design in the speech analyzing equipment. Fig. 6 shows that the addition of a fifth stage improves the frequency-response somewhat. For this reason and because a three tube twin-triode amplifier requires five coupling networks anyhow, the filter amplifiers were designed with five series-peaking networks instead of four.

Having decided upon a general filter design through a study of the low-pass filter, we must now effect a transformation in the design so that substantially the same properties will be realized in band-pass filters of selected midband frequencies. The way in which this transformation is to be made is indicated very specifically by C. V. Landon[26] in the following words: "If in a given low-pass filter a condenser is added in series with each inductance of the proper value to tune it to a frequency, f_o, and if an inductance is added in parallel with every (previously present) condenser of the proper value to tune it to parallel resonance at frequency, f_o, then the band-width of the new band-pass filter is exactly the same as the band-width of the previous low-pass filter at every attenuation ratio. In the band-pass filter the frequency, f_o, is the geometric mean of any two frequencies of equal attenuation." One additional fact should, however, be stated about the time-responses of the two filters. If a voltage of mid-band frequency is suddenly applied to the input of the band-pass filter, the envelope of the output current will be of the same form and extent as the output current of a corresponding low-pass filter when a dc voltage is suddenly applied to its input, provided, however, that the band-width of the low-pass filter is equal to not the full but the half band-width of the band-pass filter.[27]

We have already decided that the band-widths of the analyzing filters should cover equal frequency intervals below and equal pitch intervals above 1000 cps, but have not fixed their absolute magnitudes. For a decision on this point we refer to pertinent data on the time properties of the ear. The investigations of Bürck, Kotowski, and Lichte[28] show that the time required for the ear to determine the pitch of a tone in the middle audio frequency region is about ten milliseconds, also that the ear can distinguish between successive and simultaneous onsets of two tones in this frequency region, provided the interval between the successive onsets is ten milliseconds or more.[29] Data on the relation between duration and loudness are less useful because it appears that loudness goes up very rapidly after the onset and then continues to rise more slowly up to about a quarter of a second. According to data published by Munson,[30] the loudness level of a sound in the middle audio frequency region rises to within 20 db of its maximum value in 5 or 10 milliseconds.

From these hearing data we may assume that the time-response of the filters should be such that the output current will rise to within 2 or 3 db of its final value in 10 millisec. Referring to Fig. 8, we see that in the proposed filters this level is reached when $\omega_c t$ is equal to 3.3 radians. But until $\omega_c t$ is equal to 0.5 radians, there is practically no response. This interval may therefore be regarded as a time delay in the signal that may be ignored. Hence the effective transition interval is about 2.8 radians. If this transition is to occur in 10 millisec., then ω_c must be equal to 280 radians per sec. and $f_c = \frac{\omega_c}{2\pi} = 45$ cps. This would be the band-width of a low-pass filter having the specified time-response. The band-width of the corresponding band-pass filters should be twice this value or about 100 cps.

Eight filters, covering the range from 300 to 1000 in 100 cps steps, have been designed and constructed in accordance with the preceding general specifications. The several cathode resistors were adjusted for zero gain at mid-band frequency in each stage. The purpose of this high feed-back adjustment was two-fold, to obtain a high plate resistance and low distortion. The measured frequency-responses of the 300 and 1000 cycle filters are shown in Fig. 9. The theoretical frequency-responses of corresponding band-pass filters derived from a low-pass Gaussian filter are indicated by the dashed lines. The response of one of these filters to a 50 db increase and decrease in the input level, as obtained with the level recorder are shown in Fig. 1[illegible]-I. A comparison of these curves with those of Fig. 3a and 3b shows the much superior transient performance of the Gaussian type of filter.

The Level Recorder

In the schematic drawing, Fig. 1, of the analyzer, each filter is terminated in a recorder. The prime function of these recorders is the synchronous registration of the time variations in the power flowing through the several filters. Since the band-widths of all the filters transmitting below 1000 cps are the same, the operating requirements for the associated recorders are the same also. We shall briefly review these requirements, leaving the recording problem in the higher frequency bands for consideration in the next section.

We have already noted that the recorder should have a logarithmic power scale with an operating range of at least 60 db and that it should register correctly the power level of waves having peak-factors at least as high as 10 db. Referring to Fig. 8, we see that, when the input voltage of a four stage analyzer-filter is suddenly increased by 50 db, the maximum rate of increase in the output power will be about 30 db per radian. The corresponding rate for one of the five stage analyzer filters would be nearly the same, as may be seen by comparison with the curve in the same figure for a Gaussian filter, which is the limiting form when the number of stages is increased indefinitely. Since the band-width of the analyzer filters is 100 cps, the cutoff frequency, ω_c, is equal to 100π radians per sec. The build-up rate in db per sec. is therefore $30 \times 100\pi = 9{,}500$. Similarly we find by reference to Fig. 8 that the corresponding decay rate is about 6500 db per sec. The recorder should thus be able to follow increasing and decreasing level changes at rates of about 9500 and 6500 db per sec., respectively.

The record should preferably be produced in rectangular coordinates with time as the independent variable and in immediately readable form. It should be permanent and not smudge in handling.

A consideration of the various possible types of level recorders led to the conclusion that these requirements could be most easily satisfied in a servo-operated device with an angularly deflecting moving coil drive of the type used in ordinary dc ammeters. The design carried out on this principle is shown schematically in Fig. 10. Instead of the usual rectangular coil, a round one is used. While the rectangular coil can theoretically have a greater ratio of torque to moment of inertia for a given power input, the round coil can more easily be wound compactly to accurate dimensions and be given the rigidity necessary for withstanding high accelerating forces.

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The input leads of the recorder are connected to the outer terminals of a potentiometer. The grounded one of these terminals and the sliding contactor are connected, respectively, to the cathode and the grid of the first stage of an amplifier. The contacting brush of the potentiometer is carried by an arm which is fixed radially to the shaft supporting the driving coil. The amplifier terminates in a twin triode used as a full-wave rectifier. The rectified current is led directly to the driving coil without any ripple filter.

While there is no signal voltage on the input, a biasing current flowing through the coil from a source in the control circuit forces the coil against a stop. When a voltage is applied to the input terminals, the resulting rectified current flows through the coil in a direction opposite to the biasing current. As soon as the former current exceeds the latter, the coil will move until enough attenuation is introduced by the potentiometer to reduce the net current through the coil to zero. The potentiometer is of logarithmic design so that equal displacements of the brush arm produce equal fractional changes of grid potential. The angular position of the coil can thus serve as a measure of the input voltage level in db.

The potentiometer is of the commutator type of construction with 100 steps of one db each. The bars are wedge-shaped and made of silver. The taper of the wedges is such that, when the bars are assembled with sheet mica separators to form the commutator and this is placed in position in the recorder, the sheets of mica, if extended, would each pass through the axis of rotation. The brush does not ride on the cylindrical surface of the commutator as in the conventional type of dc motor but on a lateral face whose plane is perpendicular to the axis of rotation. This face was finished by mounting the commutator rigidly on the face plate of a lathe and taking light cuts with a wide, hard, and sharp tool. This manner of finishing left no noticeable burrs at the edges of the silver bars and produced a smooth contacting surface without observable ripples, a prerequisite in a potentiometer designed for high speed operation. The commutator is mounted with its face close to the brush arm. This arm is a metal conical shell, a form that gives it great rigidity with a low moment of inertia. The brush consists of a pair of thin berillium-copper cantilever springs with their bases soldered to the brush arm and the tips resting on the commutator. These tips are bent into a small U in order to reduce the danger of chattering. One spring is slightly longer than the other so that the brush spans the insulating mica separators between the commutator bars. An enlarged view of a portion of the face of the commutator and the end of the brush arm with the attached brush is shown in the lower right hand corner of Fig. 10.

A prime objective in the design of the potentiometer was chatter-free operation at low contact pressures. A potentiometer constructed as here described was used in the recording of all the speech data presented in this report. Good contact was maintained between brush and commutator at the highest operating speeds with a contact force of not more than a gram in each spring. No lubricant was ever used, yet the many hours of operation produced no detectible erosion on either commutator or brush. Now and then poor contact developed from an accululation of dirt. Silver sulphide had occasionally to be removed from the face of the commutator with a mild silver polish.

In addition to the brush arm, the coil carriage supports an arm for moving a writing pen. This arm is in the general form of a truncated, thin, conical shell. It is attached at its basal end to the carriage with its axis perpendicular to the axis of rotation and extends downwardly to within a half inch of the recording paper below. Fitting freely, but not loosely, into the truncated apical end of this conical shell is a tube which forms the main stem of the recording pen. This stem terminates in the pen-point at its lower end and in the ball of a ball and socket joint at its upper end. The pen-point consists of a small piece of hardened berillium-copper tubing with a rounded end and a 0.25 mm bore. It is press-fitted into the end of the stem. The length of the pen, from pen-point to the center of the ball, is made equal to the distance between the paper and the axis of rotation. The socket of the joint at the upper end of the stem is supported from a clamp by a second piece of tubing. This tubing acts as a spring by which the pen-point is pressed against the paper with a force of 2 or 3 grams. The spring tension as well as the horizontal position of the socket can be controlled by adjustment of the supporting clamp. Beyond this clamp the tubing is bent downward into the ink-well. The ball and socket joint is constructed so as to provide a clear passage for the ink at all operating positions of the pen arm. The surface of the ink in the well is kept below the lowest position of the pen-point so that drainage from the well by siphon action is avoided. The ink is held in the channel between the well and the pen-point by capillary action.

The strip of paper on which the power level is recorded is moved in a direction parallel to the axis of rotation with its center-line directly below this axis. Where the pen rests on it the paper is curved by being passed through a narrow curved slit. The center of curvature of this slit coincides with the axis of rotation so that, as the pen-point is moved across the paper, there is no longitudinal displacement of the stem of the pen. The paper is propelled through this slit by a pair of opposing narrow metal rollers placed close to the ingoing side of the slit and bearing on the paper along its center-line. One roller is driven; the other is a spring pressed idler. The paper is drawn by these rollers from a 300 foot supply roll. The lateral position of the paper at the pen is controlled solely by adjustment of the position of the supply roll along its axial direction. Movement of the paper may be started or stopped quickly by a relay through which the idler roller is raised from the paper against a spring pressure.

The total travel of the pen-point is 5 cm. While it traverses this distance the potentiometer brush passes over the one hundred steps of the commutator. A half mm displacement of the pen-point thus represents 1 db.

For some measurements paper ruled in cm squares with mm subdivisions was used but the speech records were all taken on paper having only longitudinally ruled lines spaced at 0.5 cm.

Fig. 11 is a drawing of the electrical circuit of the recorder. A 3-stage feed-back amplifier is followed by a phase inverter stage and a twin triode full-wave rectifier. E is the power supply for the coil biasing current. The two rheostats, which are adjustable by a common shaft, control the pen speed and thus the degree of smoothing of the input power. The resistance values in the potentiometers were so selected that the steady state readings of the recorder are independent of the potentiometer settings.

For the study of peak-factor errors in the recorder a peaked wave generator, schematically shown in Fig. 12, was constructed. In this generator an oscillator is connected across a resistance, R_1, a battery, E_1, and a pair of diodes poled so as to oppose current flow from the battery. The input of an amplifier is shunted across the resistance. With the battery set to a particular value a voltage output having a wave of the general form indicated in the upper right-hand corner of the figure will be developed whenever the peak value of the oscillator voltage exceeds the battery emf. The peak factor of this wave can be controlled by adjustment of the battery and oscillator voltages.

The circuitry shown in the figure to the right of the amplifier is used to measure the peak factor of the wave. A thermo-ammeter in series with a resistor, R_2, is connected across the output. Its reading multiplied by the resistance gives the rms value of the output voltage. By the closing of switch, S, a second shunt circuit is placed across the output terminals of the amplifier, comprising a small resistance, R_3, the resistance, R_p, of a voltage divider, and a diode rectifier opposing current flow from the battery, E_2. A sensitive oscilloscope is placed across the resistance, R_3, and a dc voltmeter across R_p. In the measurement of the peak voltage R_p or E_2 are

varied until the oscilloscope barely indicates a flow of current through R_3. The reading of the voltmeter will then be the peak value of the output voltage. After the peak factor of the wave has been thus determined, this measuring circuit may be disconnected by the switch, S, and the terminals a and b used as a source of voltage of known peak-factor.

For the measurement of peak-factor errors in the recorder the driving coil, with a milliammeter inserted in series, was held fixed somewhere near the middle of its scale. A sine wave voltage was first applied to the recorder and its value adjusted until the coil current was zero. The rms input voltage was then read with a thermo-ammeter connected in series with a known high resistance. Next a voltage having a peaked wave of known peak-factor, generated as described in the preceding paragraphs, was applied to the input and its level adjusted to a null coil current as before. The thermo-ammeter was then again read. The difference in level between the two thermo-ammeter readings is equal to the level error in the indications of the recorder. The recorder circuit was adjusted so that this error was negligibly small for peak factors up to at least 10 db. In this condition it was used for obtaining all of the speech data here reported.

Fig. 13 shows the writing speeds of the recorder in its final state of adjustment, as it was used for the speech analyzer. The records reproduced in this figure were obtained when a 1000 cps input voltage was suddenly increased and decreased by 50 db. The Roman numbers attached to the 8 different records refer to the corresponding speed control potentiometer settings. The numbers given alongside of the build-up and decay lines are the respective build-up and decay speeds, as derived from the slopes, in db per sec. The space between two horizontal lines represents 10 db. An equal distance in the horizontal direction represents 0.02 sec.

It will be noted that at the highest recording speeds there is a considerable amount of over-shooting, but Fig. 8 shows that when a voltage is suddenly applied to one of the filter amplifiers, with which the recorder is used for speech analysis, the output rises rapidly at first and then more slowly as the equilibrium value is approached. When this filter output voltage is applied to the recorder, the over-shoot is small as may be seen from the records shown in Fig. 14, which correspond to those of Fig. 13 except for the interposition of the 1000 cps filter between the oscillator and the level recorder.

The build-up and decay rates of the filters measured with the recorder are in good agreement with the theoretical values previously given in this section.

In the records of Fig. 13 the build-up rates exceed the decay rates. This property of the recorder is here no handicap since the filters exhibit a similar type of behavior. For recording the level of sound waves having a lower peak-factor and more nearly equal maximum onset and stoppage rates than those of speech, the recorder can be set to have high and equalized build-up and decay rates and less over-shooting. This fact is shown by the record reproduced in Fig. 15, but with the recorder adjusted to yield this curve the maximum allowable peak-factor was only 5 db.

We have considered the problem of pen speed in some detail but have barely mentioned pen acceleration, yet high recording speeds are of little avail if much time is consumed in reaching these speeds. The build-up and decay curves show that in this recorder the maximum speeds are, as a matter of fact, attained in a negligibly short time, particularly in the more important case of the build-up. A related but perhaps more serious problem is that of over-shooting, for when the recorder is set for highest pen speed, we depend upon the special filter characteristics to keep the over-shoot within acceptable bounds. It appears at present to be the limiting factor in the pen speed.

If the recorder has been properly engineered circuitwise, the over-shoot will be inversely related to the electrical damping of the moving system, which is determined primarily by the ratio B^2/I, where B is the effective flux density in the air-gap and I is the moment of inertia of the moving system. This ratio should therefore be made as high as practicable.

If a well-defined continuous inked line is to be obtained at high writing speeds, the pen and its carriage must be kept free of perceptible resonance vibrations. Because of the requirement of high structural rigidity that this condition imposes, a reduction in the moment of inertia below the 4 or 5 gm cm^2 of the present recorder looks difficult.

Increasing the magnetic field strength to improve the damping is an attractive idea because its effect is proportional to the square of the flux density, but a sizable increase above the 12,500 gauss of the present recorder might be costly and result in a bulky instrument.

A more effective procedure would be to reduce the scale factor of the recorder. A one db change in level might be presented by a deflection of 1/4 instead of 1/2 mm. This smaller scale would be no great handicap for many types of level measurements, particularly if the pen were modified so as to produce a somewhat finer line. As an illustration of what might be done let us consider the following changes: reduction in the length of the stem of the pen by one-half, the moment of inertia of the pen and its driving arm would thereby be reduced to 1/8 of its present value; reduction in the diameter of the coil by one-half while its cross-sectional area is doubled, this change would reduce the moment of inertia of the coil to 1/4 of its present value. If the same sized wire were used, the resistance of the coil, the back emf per unit angular velocity, and the torque per unit of current would remain the same. This driving unit could then be used in the present circuit and the speeds, expressed in either radians or decibels per second, would be unchanged; but the moment of inertia of the moving system would have about 1/4 of the present value, the accelerations would consequently be higher, and the over-shoot in db would not be more than 1/4 as great.

If a recorder is to be capable of following level variations that take place at a certain maximum speed, then its response to a sudden increase in input level must be faster than this speed for the same reason that the response to a sudden change in input voltage is slower for two similar filters used in cascade than it is for either one of them used separately. The amount of this required excess of speed, if the recording errors are to remain within tolerable limits, depends upon the character of the time-response of the recorder. If, for instance, this were similar in form but twice as fast as that of one of the filter amplifiers which we have been considering, then the maximum rate of level variations that would be indicated by the recorder following a sudden change in level at the input of the amplifier would still be found to be about 10% less than the actual rate of change of level at the input of the recorder.[31] On the other hand, if the operation of the recorder were such that a change in input level equivalent to one step of the input potentiometer produced the same unbalance in the coil current as would a sudden larger change in level, the required excess of speed would be practically zero. The level recorder here described comes near to operating in this latter way, for, with the recorder set just as it was used for making the speech records, a change of only six db in the input level develops the maximum torque in the driving coil.

System Design and Experimental Procedure

By deliberating upon the filter characteristics and the time properties of the ear we have already concluded that in the proposed speech analyzer the filters transmitting below 1000 cps should have band-widths of 1000 cps and those transmitting above this frequency, band-widths equal to 10% of their mid-band frequencies. If, with this stipulation, the equipment were set up strictly in accordance with the schematic drawing of Fig. 1 and if the frequency range from 250 to 6500 cps were covered by filters with adjoining transmission bands, more than 25 filters and recorders would be needed. In order to avoid assembling this mass of equipment we followed Sacia's[12] procedure of making a record of the speech sound to be analyzed on a belt loop and then reproducing it, not simultaneously through the whole bank of filters, but successively through a single filter of which the mid-band frequency was shifted the appropriate amount before each repetition. The filter amplifiers are very adaptable to this scheme so long as the band-widths are fixed, for in this case a shift can be made, without any alteration in filter characteristics, from one mid-band frequency to another, by a change only in the tuning coils and capacitors of each stage. Fig. 16 shows how this is done in one coupling network by a two pole rotary switch. A similar switch was provided for each of the other four networks of the filter. All five switches were operated from one common shaft. The fixed circuit elements, R, L, and C in each network were set for a Q of 0.56 and a band-width of 100 cps. The inductors and capacitors controlled by the gang switch were fixed at values that permitted setting of the mid-band frequencies at the hundreds from 300 to 1000 cps.

Above 1000 cps the band-widths should vary proportionally with the mid-band frequencies. The circuit elements L, C, and R, as well as the tuning reactances, would therefore have to be different for each mid-band setting. This added complexity would not leave much advantage in the variable filter over a complete set of fixed filters. The analytical data on speech above 1000 cps are therefore obtained more simply by the use in reproduction of a fixed filter and a variable record speed instead of a fixed speed and a variable filter. The analyzer was constructed on this basis. The filter-amplifier described above, set for a mid-band frequency of 1000 cps, was used as the fixed filter. If now, for example, we should with this equipment wish to determine the speech power in a band having a mid-band frequency, f_o, and a width, $0.1f_o$, we would reproduce the record at a speed, $S_p = \frac{1000}{f_o} S_r$, where S_r is the recording speed. Then the power transmitted by the filter would be a measure of the power of the original speech wave in the band extending from $.95\ f_o$ to $1.05\ f_o$.

The relation of the power indicated by the level recorder to the actual power in the original speech depends upon the frequency-response of the complete system and also upon the voltage developed in the recording head as a function of the record speed in reproduction. This problem will be discussed further in later paragraphs.

The recording medium was a loop made from a ribbon of Vicalloy 0.005 cm by 0.125 cm. This was run as a belt on two crowned pulleys. In the recording one of these pulleys was driven by a quiet, smooth-running induction motor through stepped pulleys and steel belts .005 cm thick and .125 cm wide. These

steel belts formed virtually a rigid coupling between the motor and the record medium. The linear speed of the record was about 50 cm per second. The total length of the loop was 90 cm so that speech sounds lasting about one and a half seconds could be comfortably recorded on the loop.

In the reproduction the record was again driven by a similar motor but the mechanical coupling between the motor and the record had, in addition to the steel belting, a variable speed drive. This drive comprised a pair of similar truncated hardened steel cones mounted horizontally with opposite orientation, axes parallel, and small clearance. When a steel ball is placed in the valley formed by these two cones, then, if one of the cones is turned, motion will be imparted to the second one through the ball. The axis of rotation of the ball will be parallel to the axes of the cones. The transmission is without sliding friction; it consequently adds only a small load. The transmitting speed is varied by moving the ball along the valley. This combination gives a smooth transmission, but the maximum load that can be transmitted by it is rather small since, as described, the contact pressures are derived solely from the weight of the ball. For larger loads the ball can be fitted with a shaft with its axis in line with the axis of rotation through which an additional force can be exerted and the load capacity increased. The range of speed ratios obtainable in the drive as it was constructed is 6.5 to 1.

The frequency modulation experienced in the reproduced sound as a result of the combined actions of the recording and reproducing drives was negligible as determined by instrumental observations on a recorded and reproduced oscillator current of 6000 cps. The principal flutter component, which had a frequency of 60 cps, was introduced by the motor itself.

The biasing current used in recording had a frequency of 100 kc and was filtered to a high degree of purity. The same head was used for recording and reproducing. Its performance was exceedingly stable. Variations in the combined recording and reproducing response during the course of the measurements, extending over the greater part of a year, did not exceed a few db at 6000 cps, provided the face of the pole-pieces and the tape were kept clean. The pole-pieces showed practically no wear at the end of this period.

The type and magnitude of the non-linear distortion introduced by the recording process is depicted by Fig. 17. The data for the curves there shown were obtained after the response of the system had been equalized in the frequency range extending from 200 to 1000 cps. With the recording biasing current set to a value intermediate between the optima for high and for low frequencies, a 300 cps current was recorded and then reproduced successively through the 300, 600, and 900 cps filter amplifiers. The output level in each case was noted. From these readings the relative strength of the fundamental, and the second and third harmonics were determined. These measurements were repeated at various values of recording current and the results plotted in Fig. 17.

If the speed of the record carrier were held to a fixed value, both in recording and in reproducing, then the difference in level between the output of the reproducing amplifier and the input of the recording amplifier added to the free field calibration of the microphone should be the same for all frequencies of importance in speech; but since in our plan of analysis the reproducing record speed is varied inversely with the recorded frequency above 1000 cps and since a magnetic tape that carries a record of a particular wave length develops a voltage on reproduction proportional to speed, this quantity should be the same up to a 1000 cps and increase at the rate of 6 db per octave above this frequency. The dots of Fig. 18 show overall response measurements made with the tape speed equal to 50 cm per sec. in reproducing as well as in recording. The free field calibration of the moving coil microphone[32] which was used in obtaining all the speech records presented in this article is included in the response calculations. A mean horizontal line connecting with a line of 45° slope at 1000 cps was drawn through the plotted points. The maximum departure of any observed point from this line is 1-1/2 db. This amount of error was deemed tolerable. If a need should arise for greater precision, any data taken with the system could be corrected in accordance with this calibration.

A schematic layout of the electrical components and their interconnections used in the recording of the speech sounds is given in Fig. 19. Not shown, however, is the 30 kc oscillator and the erasing head to which its output can be connected whenever the record belt is to be cleared of any remanent magnetism.

When a record is to be made, the switch, S_1, is closed, the microphone is thus connected to the input of a variable gain amplifier, and S_3 is thrown to the right, the output of the amplifier is thereby connected serially to the 100 kc biasing current generator, the recording head, and a small resistor shunted by a cathode ray oscilloscope. No biasing current is delivered by the generator while its plate supply, E_p, is disconnected by the relay switch, S_5. The oscilloscope was included in the circuit for monitoring purposes. Two horizontal lines were marked on the screen of the oscilloscope, one below the undeflected position of the scanning spot and the other an equal distance above it. The sensitivity of the oscilloscope-resistor combination was adjusted so that a sinusoidal current having a peak value of one milliampere would deflect the spot up to these lines. The

amplitudes of the harmonics that are generated under these conditions are indicated by the dotted vertical line in Fig. 17. In the recording of the speech sounds the peak values of the waves appearing on the oscilloscope screen were never allowed to go beyond either line. Serious overloading of the tape was thus avoided. The words to be recorded on a particular record belt were repeated several times and the amplifier gain adjusted to optimum value just before they were actually recorded. The switch, S_4, was then closed in order to short-circuit the head and remove further the danger of recording any foreign disturbances, and the tape given a final erasing. Next S_6 was closed and current set up in the relay winding, pulling the contacts to the bottom position. This operation started the flow of biasing current through the head by connecting the plates of the tubes in the oscillator to the power supply, E_p. It also removed the short-circuit from the head, and gave a signal to the speaker, through the signal light, ℓ, that the equipment was set for recording. As the capacitor, C, became charged, the current through the relay weakened until it no longer could hold the relay armature against the force of the restoring spring. When that happened, the head was again short-circuited, plate supply was disconnected from the oscillator, and the signal light was extinguished. No current could flow through the head until the capacitor was discharged by the momentary closing of switch S_7. The recording time interval could be controlled by adjustment of C or R_1. Before the record was removed or analyzed, it was reproduced over the loud-speaker, L.S., and aurally tested for any obvious faults by opening S_1 and S_4, throwing S_2 to the right and S_3 to the left.

Originally included in the recording equipment was a pulse generator, by which a short broad-band pulse was recorded directly after the switch S_6 was closed. This recorded pulse was to serve as a time mark for lining up the level records obtained for the various frequency bands of any given sound record. It was found that this extra equipment was unnecessary because, when the switch S_6 was closed, a pulse having a sharp build-up in all frequency bands was developed in the amplifier by induction. This pulse was therefore used as the time mark. All the speech records were made with the speaker in the anechoic chamber at Murray Hill, N.J. The microphone was placed at a distance of 12 or 15 inches from the speaker's mouth.

There are about 40 standard speech sounds in the English language. The exact physical structure of any one of these sounds will depend not only upon the speaker but also upon the sounds that precede and follow it, i.e., upon its phonetic environment. In order to get a proper concept of the general physical nature of these sounds we need therefore to study them in relation to their environment; but obviously consideration of every possible environment of every speech sound is impracticable. We therefore had the problem of selecting for presentation a limited amount of successively through the 300, 600, and 900 cps filter amplifiers. The output level in each case was noted. From these readings the relative strength of the fundamental, and the second and third harmonics were determined. These measurements were repeated at various values of recording current and the results plotted in Fig. 17.

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The records from which the charts labelled A, B, C, and D on page 100ff were derived were made respectively by Gordon E. Peterson, Ph.d., Associate Professor, Speech Department, University of Michigan, dialect: General American; Malcolm S. Coxe, Ph.D., Assistant Professor of Speech, Brooklyn College, dialect: Southern and General American; Mary B. Coxe, M.A., Speech Therapist, Brooklyn College Speech and Hearing Centre, Brooklyn College, dialect: General American; Annette Zaner, M.A., Speech Therapist, Brooklyn College and Beth Israel Hospital, dialect: Eastern and General American.

For the analysis of the speech sounds recorded on the tapes the driving shafts of the tape-loop and of the paper strip in the level recorder were rigidly coupled through a steel belt. The linear speed of the paper was set at one-half the tape speed. The arrangement and the electrical interconnections of the principal parts of the equipment are shown in Fig. 19. The switches S_1 and S_4 were left open, and S_3 was thrown to the left. For the analysis of the records on a particular belt, S_2 was first connected to the loud-speaker for an aural check on the identity and quality of the records. After this test S_2 was thrown to the left. Thus the variable filter, followed by the level recorder, was connected to the output of the amplifier. A calibrated oscilloscope was connected across the input of the filter. It provided a means for making certain that the input voltage did not overload the filter. Next the filter setting giving the highest output level reading was found by test and the gain of the amplifier set so that the level reading for this particular setting was about 60 db. Level-time records were then made at the eight different filter settings with the tape running at the recording speed; afterwards the filter was set at 1000 cps and the speed reduced in steps of about 10% down to one-sixth of the recording speed. A level record was made at each step. In this way a level-time graph was obtained for each of twenty-six adjacent frequency bands of equal width on the Koenig scale.

When a complex tone is applied to the input of a band-pass filter whose transmission band is wide enough to embrace a number of harmonics, the power output of the filter will have a periodic component with a fundamental frequency equal to that of the tone, except for the case, unlikely in speech, where the sound contains no even harmonics. The records of the voiced portions of speech exhibit this periodicity more and more prominently with increasing band frequency from 1000 cps on upwards, partly for the reason that the bands by becoming wider transmit a larger number of harmonics and partly because the reproducing speed is slowed down so that the level recorder can follow these periodic variations more closely. The magnitude of this periodic variation in power depends upon the amplitudes and phases of the harmonics in the tone. Hence it is possible that, because of a slight difference in the phase relationship, the periodic variations in neighboring bands may differ more than one might expect. Not too much value can therefore be placed on the appearance of these periodic variations in the level record. It shows whether the corresponding speech sound was voiced or unvoiced and, if voiced, what its pitch was. On the other hand, the periodic pattern obscures the level of the mean value of the power, a knowledge of which may be important in the specification of the quality of the sound. Wherever the periodic level variations were of high amplitude, a second record was therefore made with the recorder set for low speed operation thus giving the levels of the power averaged over a longer time interval. The record so obtained was traced over the first one with a pencil. These traced-in average level values are in most cases easily identified in the printed graphs.

For the assembly of the records the strip of paper on which they were inscribed was trimmed down by a special slitter close to the longitudinally ruled base and 60 db level lines. It was then cut into sections so that each piece had on it the level-time record in one particular frequency band of one particular belt, usually two words. The sections belonging to any one belt were pasted on a card-board, one above the other in the order of increasing frequency and with the time marks in vertical alignment. With a ball-point pen other lines were then drawn parallel to the vertical row of time marks and spaced at a distance corresponding to 0.02 seconds in the recorded speech sounds. The cards were then cut apart vertically in order to contain all the level-time information on one word only. The printed charts presented on pages 101 to 220 inclusive were made from these cards at a reduction in size of three and three-quarters to one. The numbers at the top of the charts are decibel values of attenuation in the amplifier. The first one of these shows the attenuator setting at which the record was made and the second one, the setting at which it was reproduced.

In the case of the ten vowels average level values for each frequency-band were read from the records. The time intervals over which the averages were taken are indicated by the arrows near the bottom of the charts. These average values are plotted against mid-band frequency in Fig. 20. The level-scales associated with the graphs in these figures have been adjusted to a common basis for all the spectral plots so that equal level indications in a particular frequency band mean equal speech powers within that band.

The vowels from which the spectra shown in Fig. 20 were derived were spoken in a conversational manner. An inspection of the level time charts of these vowels on pages 100 to 110 reveals a considerable variation in level in many of the frequency bands during even the relatively short time over which levels were averaged in deriving the spectra. Because of the difficulty of determining the proper average values in these cases, a new set of vowel records was made, six or eight months after the first set, for which the speakers were asked to say the vowels in a sustained manner, although some of them expressed the view that the short vowels could not be sustained without doing violence to their characteristic qualities. From these records the spectral curves of Fig. 21 were derived in the same way as those of Fig.20. The most interesting fact to be noted here is that corresponding curves of Figs. 20 and 21 are nearly alike even to most of their finer details. Records were also made of sustained semi-vowels and unvoiced fricatives. The spectral charts derived from these records are given in Fig. 22.

General Remarks

The objective of this work was the presentation of quantitative experimental data on the sounds of speech under typical phonetic environmental conditions and in a form that would be most representative of the way in which these sounds are perceived and analyzed by the ear. It should offer study material for anyone who is interested in the scientific or engineering aspects of the subject. An interpretive discussion of the speech charts is therefore outside of the realm of our objective, but there are a few special observations that we should like to make.

Some of the records show spurts or spikes of power in rather unexpected places. Corresponding spikes generally appear in most of the bands. This spreading of the power in frequency indicates that the spurts are of short duration. They are not faults in the record material, at least not in most cases, but incidental noises that are generated in the mouth of the speaker. In some records these spikes occurred in places where it was difficult to tell whether or not they were part of the speech sound. In these cases the particular speech sound was re-recorded. It is believed that in the remaining records wherever these spikes occur they come in parts of the record where they are not easily mistaken. For this reason and because they illustrate the typical character of some of the noise that accompanies the production of speech, these sounds were not recorded.

An objection often raised against the resonance theory of hearing is that some of the properties of the ear, such as the great speed with which it can determine the pitch of a tone, would demand a highly damped resonance system; while other of its properties, such as its ability to sense small changes in pitch, would demand a system of low damping. Since these requirements seem to be incompatible, it has sometimes been assumed that the ear accommodates its effective damping to the requirement of the moment. However, the analyzer here described in its unmodified form, i.e., in the form in which the 26 filters and recorders are embodied physically so that they are all in action simultaneously, is in an operational sense an artificial ear. It has no feedback scheme of tuning adjustment, yet it has a speed of response in every respect comparable with the speed of perception of the ear and it can determine pitch with comparable accuracy if the response characteristics of the filters are accurately known and the recorders are sufficiently sensitive. The pitch of a pure tone is determined by the level reading of the two channels having the highest output. When there are several harmonic components of low order, the pitch is determined by the output level readings of a proportionately larger number of channels; the precision with which the pitch may thus be determined is ultimately limited only by thermal noise just as it is for the ear in its most sensitive frequency region. If the tone, like most vowels, contains mainly harmonics of higher order, then the pitch can be determined rather precisely by the frequency of the periodic variations in level exhibited in the upper bands. This last method of determining pitch corresponds to that which for the ear J. F. Schouten has termed the residue.[33]

We shall not attempt to set up specific rules for the identification of the various speech sounds by their level-frequency-time records. While the vowels, semi-vowels, and the un-voiced fricatives are identified mainly by their spectrum envelopes and by the appearance of their level-time records, whether periodic or aperiodic, it is obvious from a study of the charts that other speech sounds are not always represented in one particular way. There are generally several factors that have a bearing on the identification, some or all of which may play a part in any particular instance. For example, in the case of the plosives some of these factors are: spikes, representing clicks, which at a given instant may occur in varying amplitudes throughout the frequency bands; steepness and character of the build-up and decay curves of the following and preceding sounds; and variation in the time of onset of the plosive within the array of frequency bands.

One can easily distinguish between the tonal and atonal sounds because the former exhibit periodicity in the power pulses, which are observed particularly in the higher frequency bands.

Some voiced sounds exhibit both tonal and atonal qualities at the same time. In these cases the bulk of the power of each portion comes in different frequency regions so that it is possible to study the character of each portion separately.

The time and frequency pattern of a sound may change considerably with its phonetic environment, and, of course, reciprocally the environment may be influenced by the particular sound. Part of the information used by the ear in the identification of the sound is then contained in the preceding and following sounds, in some cases perhaps even the greater part of the information. These are facts that have long been recognized by phoneticians. Indeed, some people may be surprised to find as much of a break between successive sounds in speech as the records show, particularly in the higher frequency bands.

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INDEX TO CHARTS

TITLES FOR FIGURES

Titles for Figures - 2

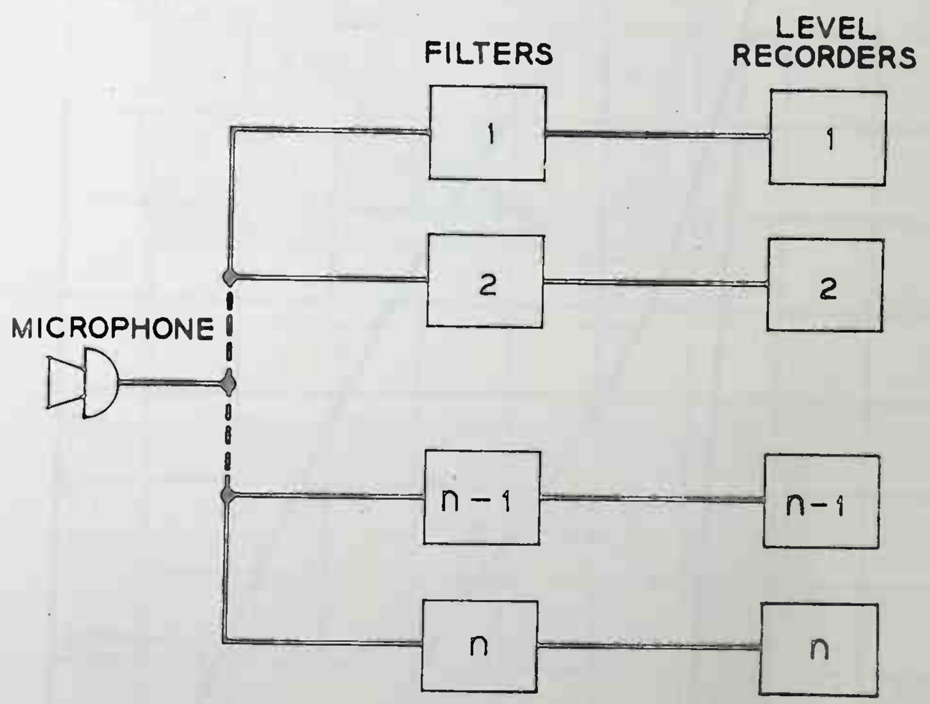


FIGURE 1

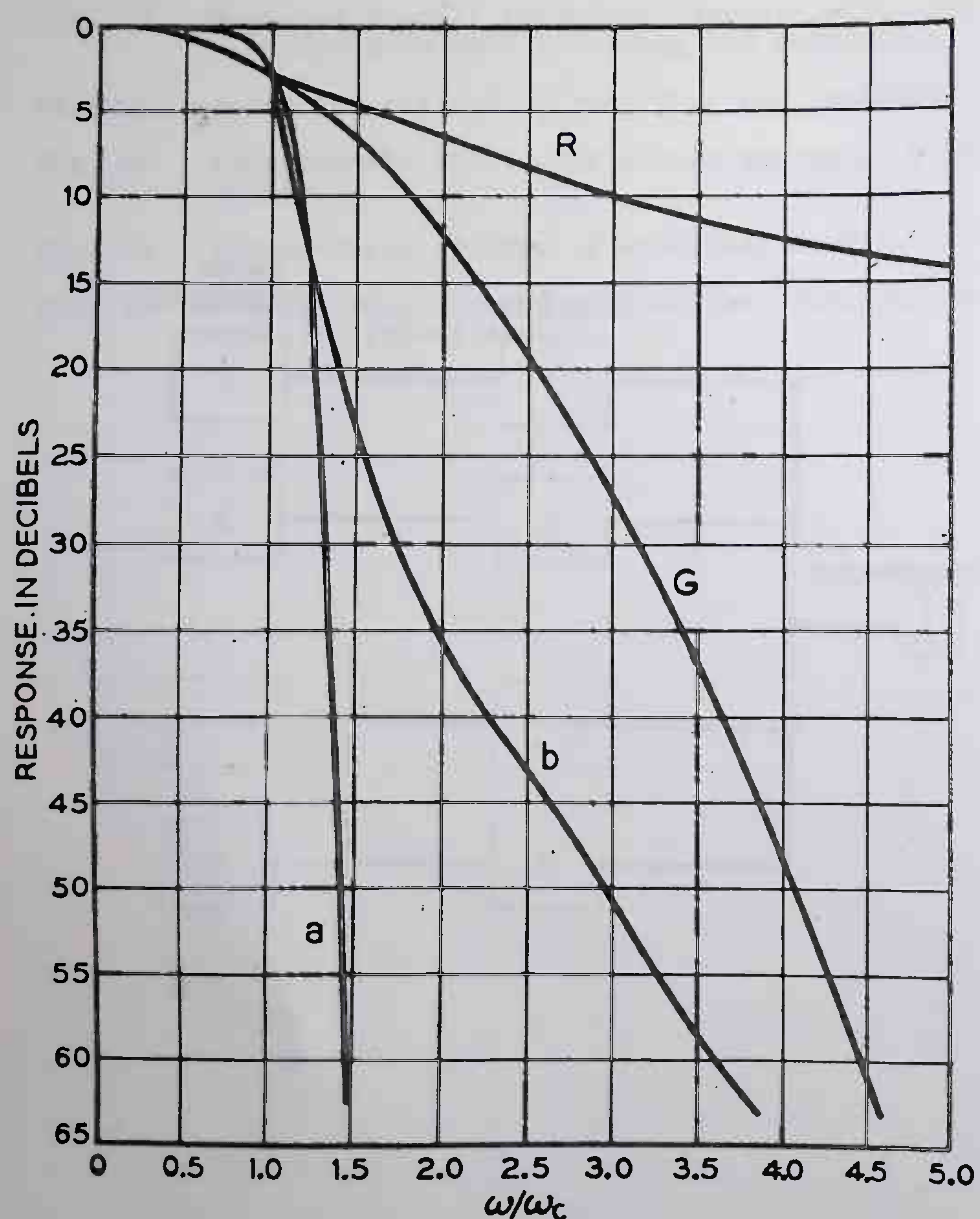
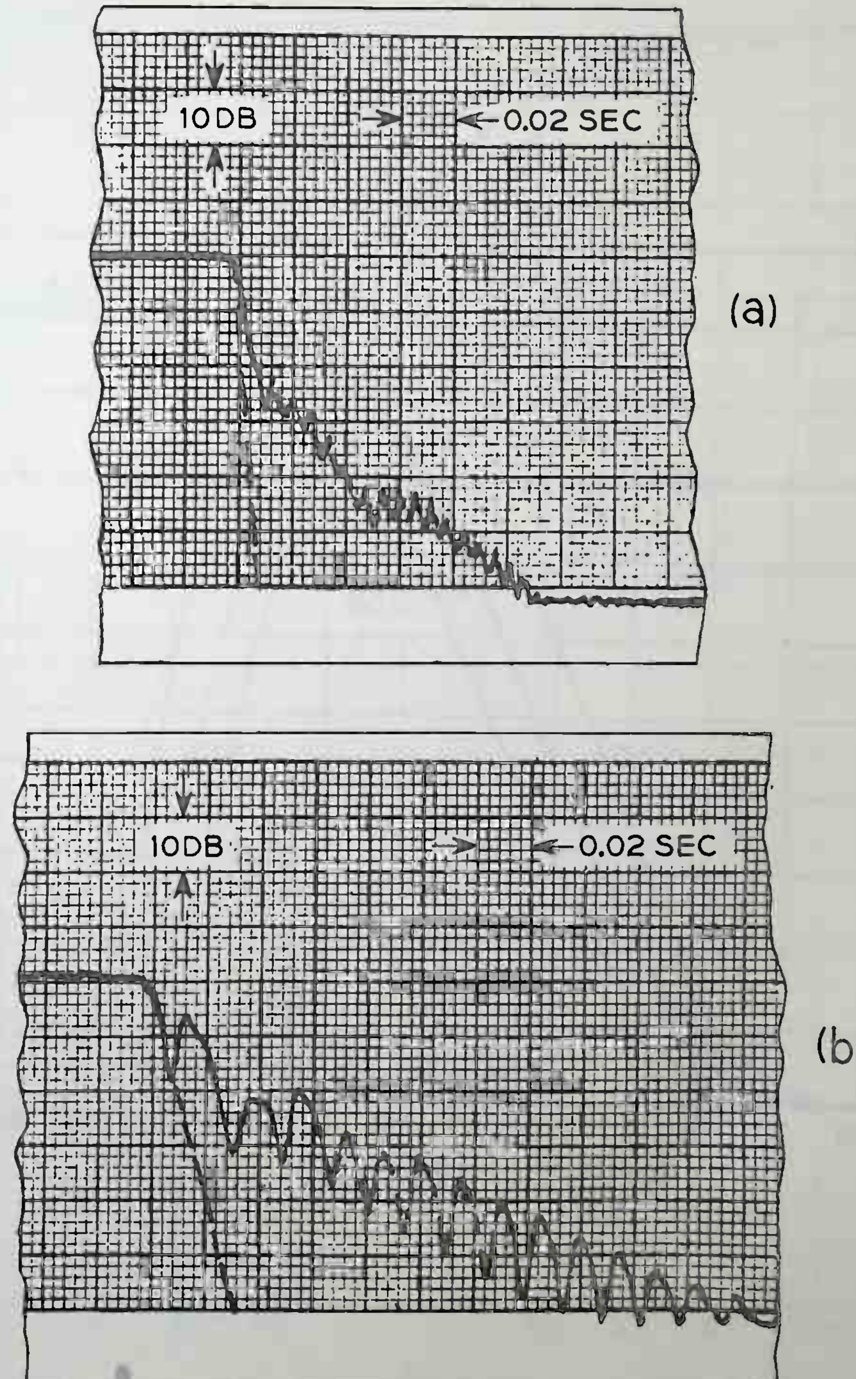


FIGURE 2



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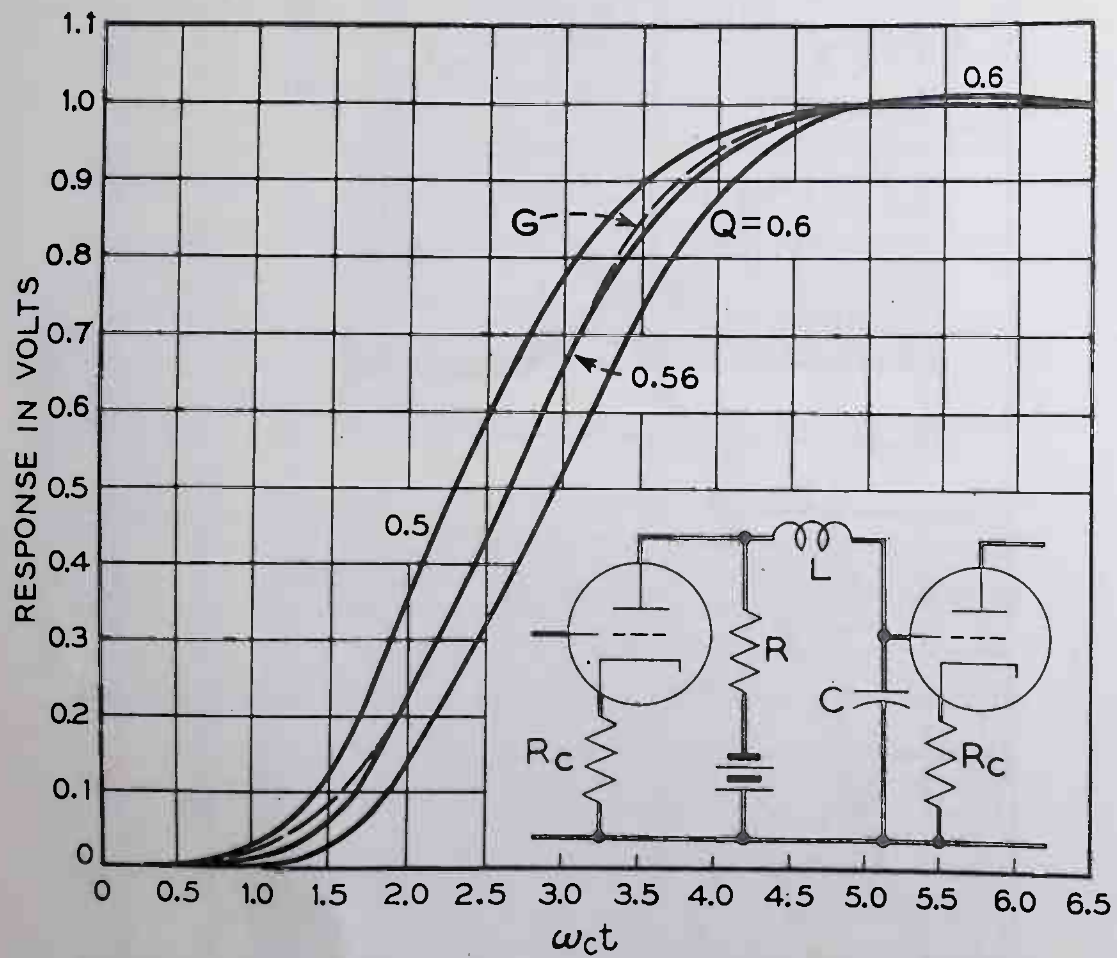

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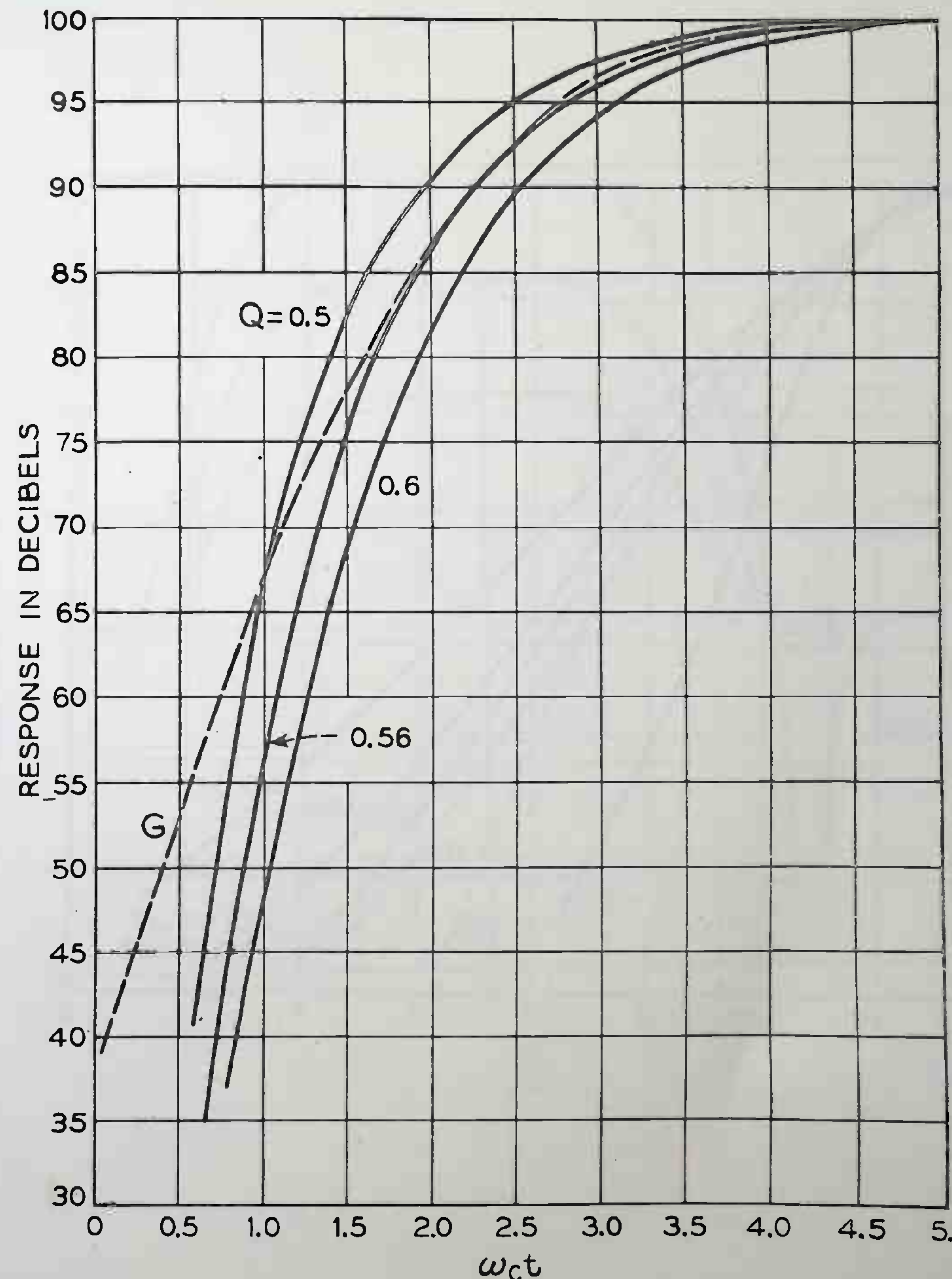

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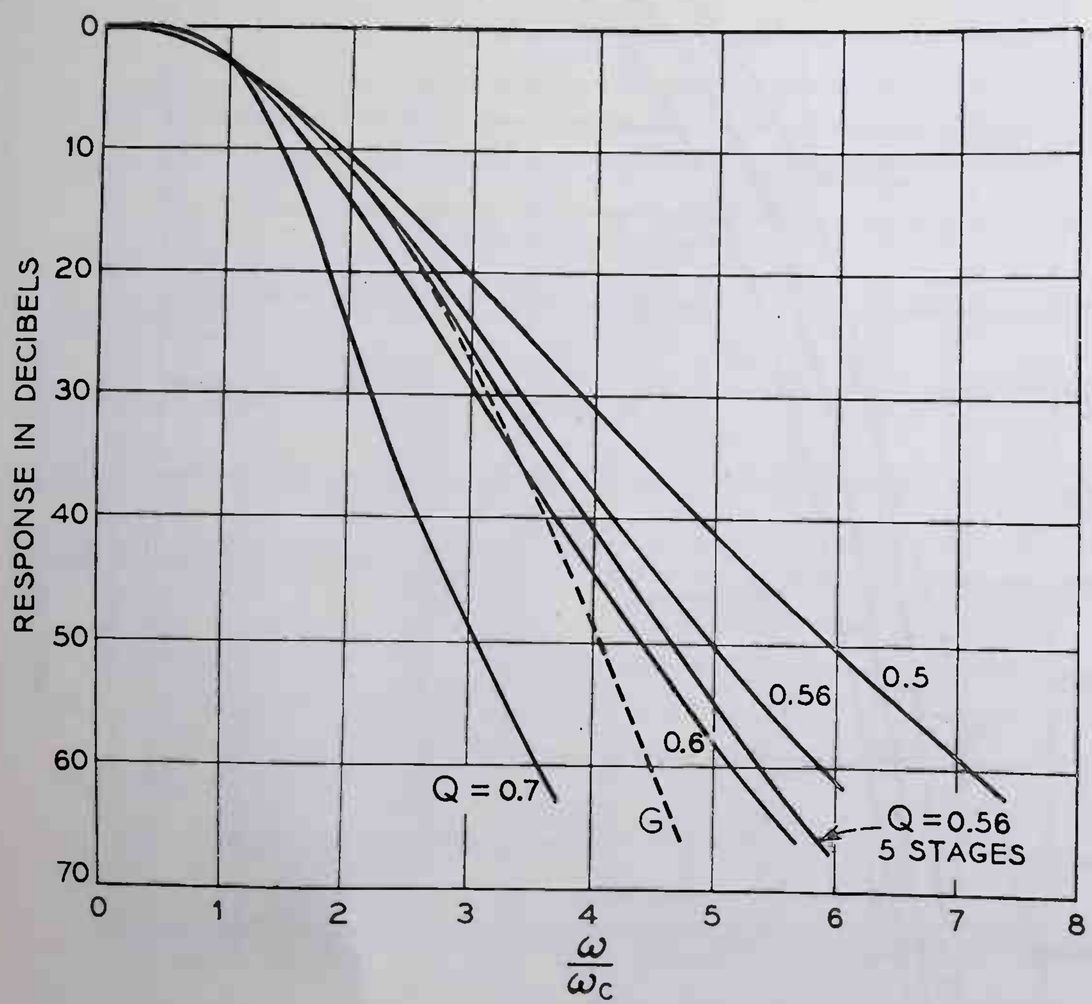


FIGURE 6

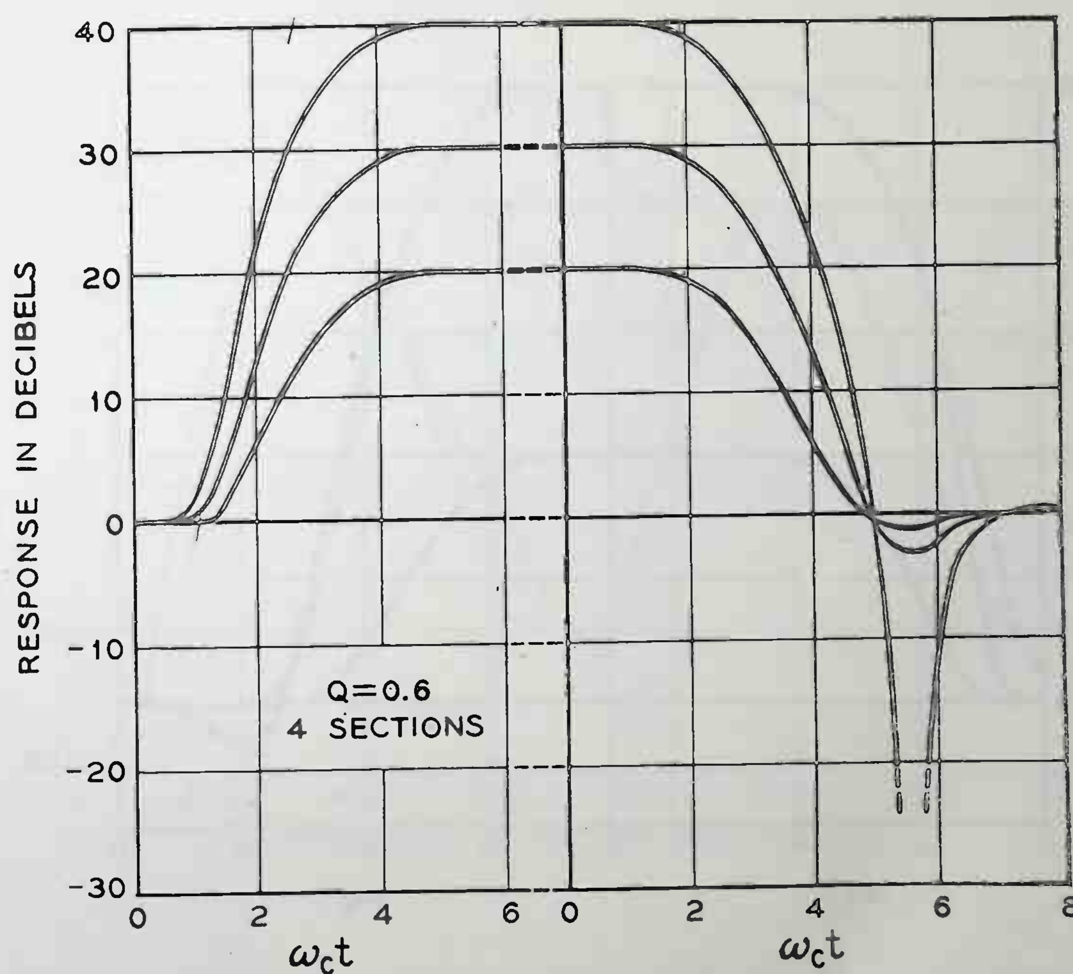
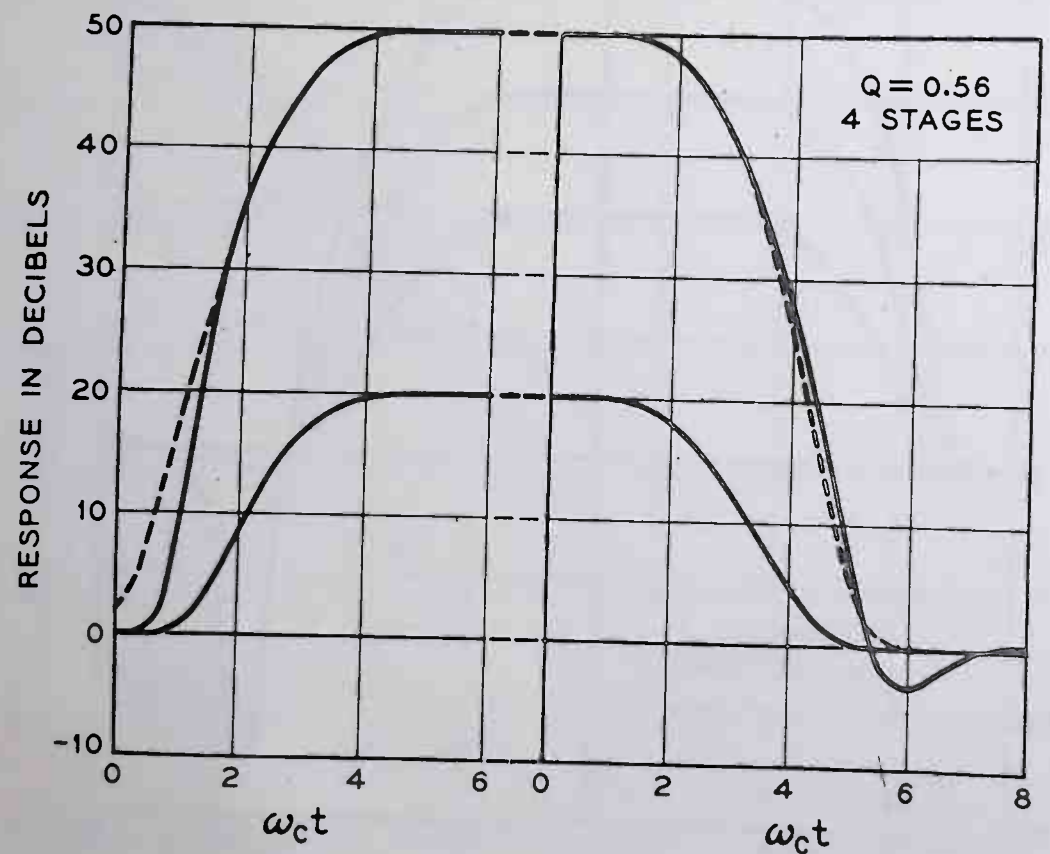


FIGURE 7



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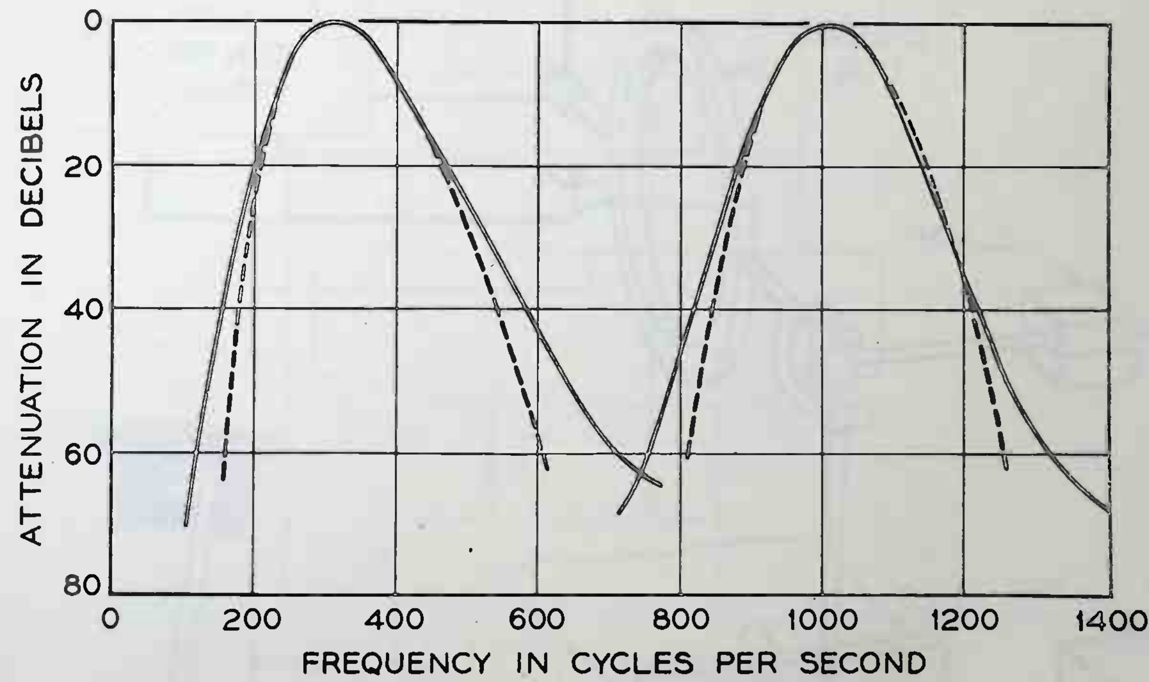


FIGURE 9

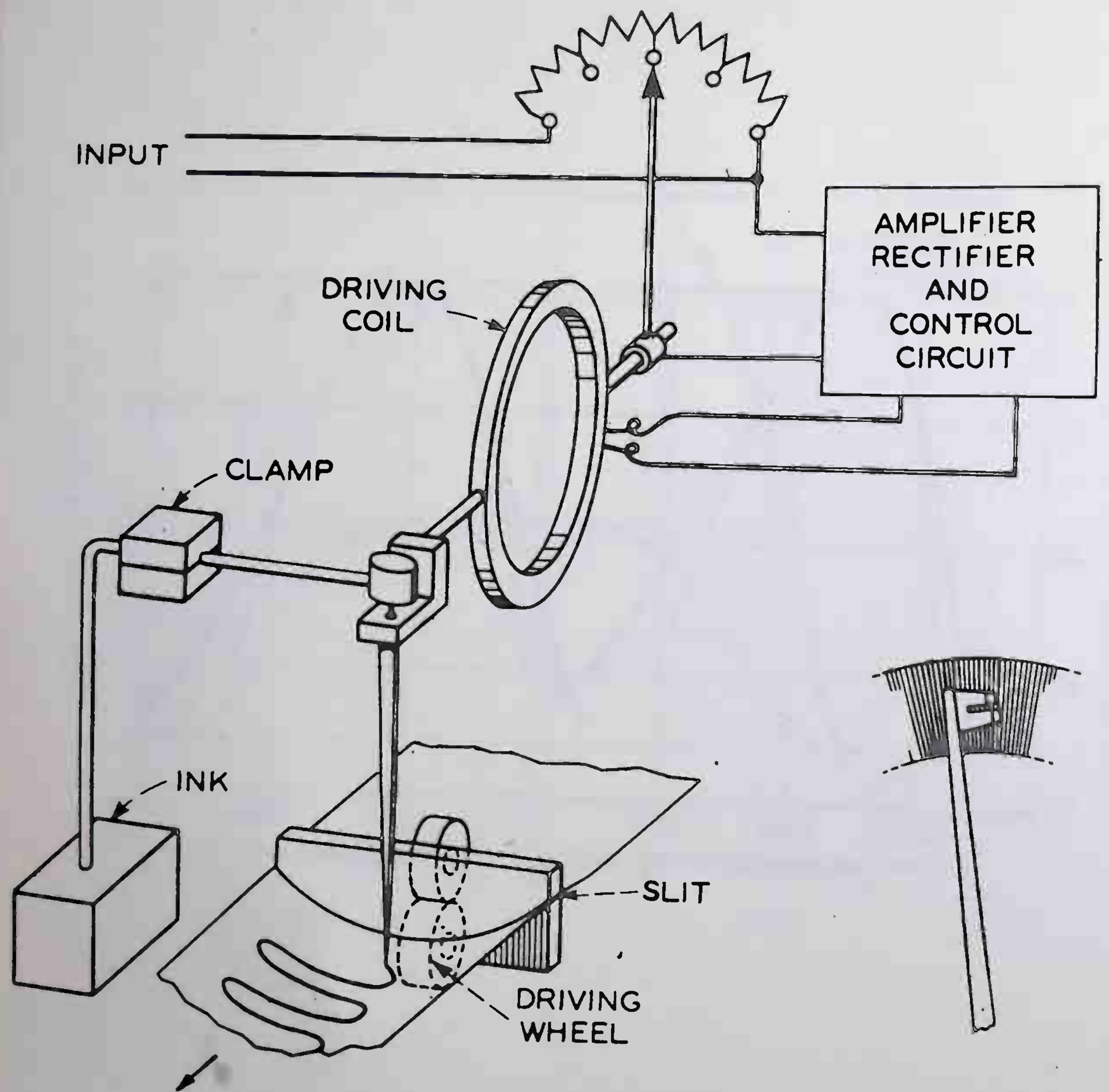


FIGURE 10

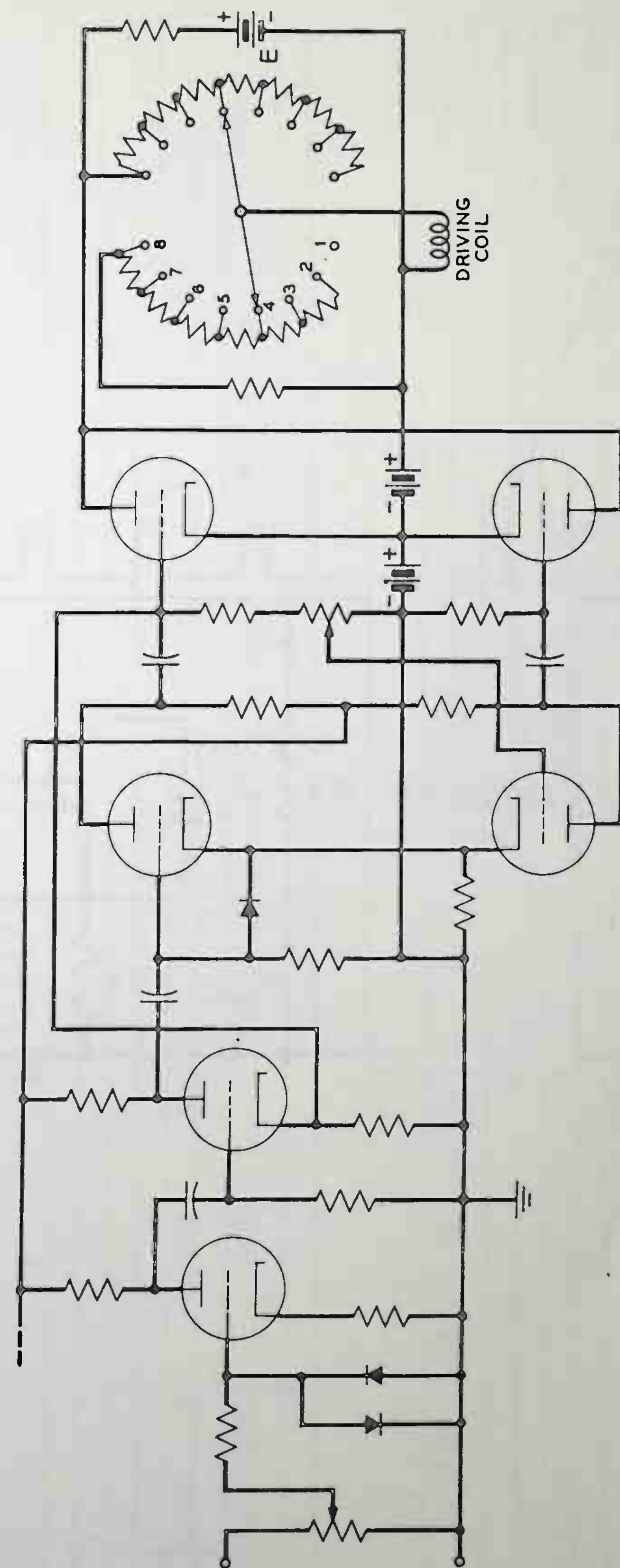


FIGURE 11

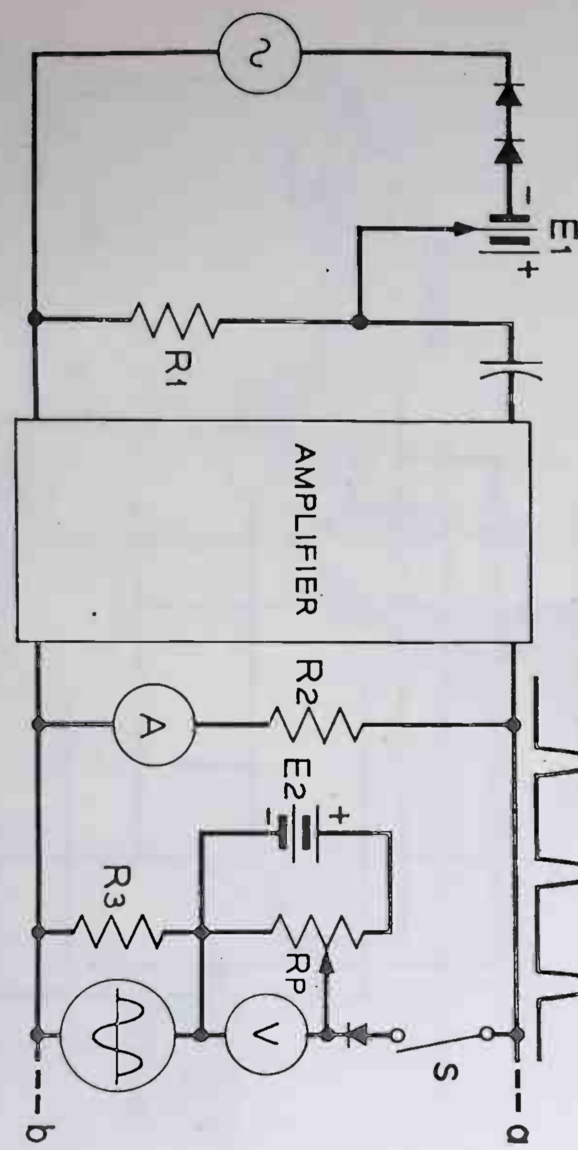


FIGURE 12

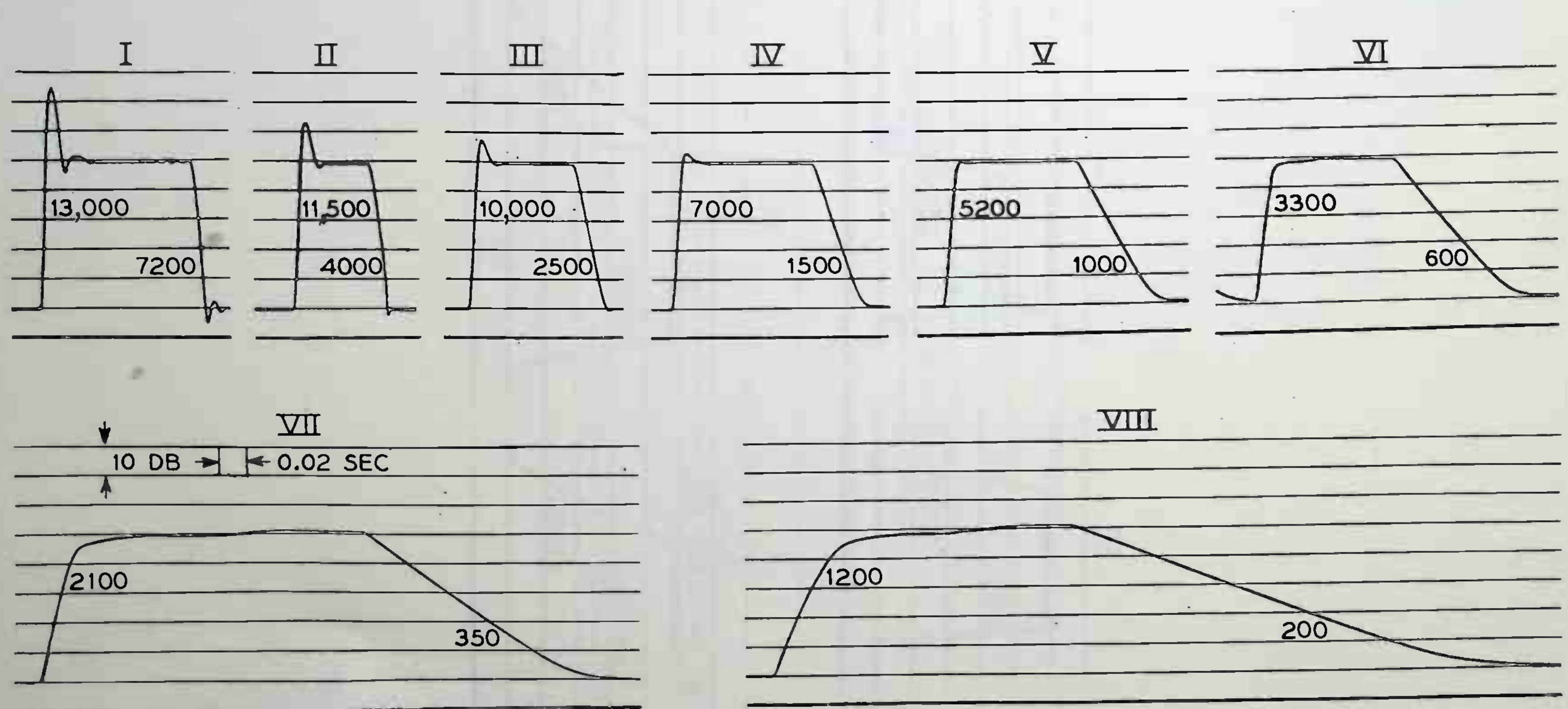


FIGURE 13

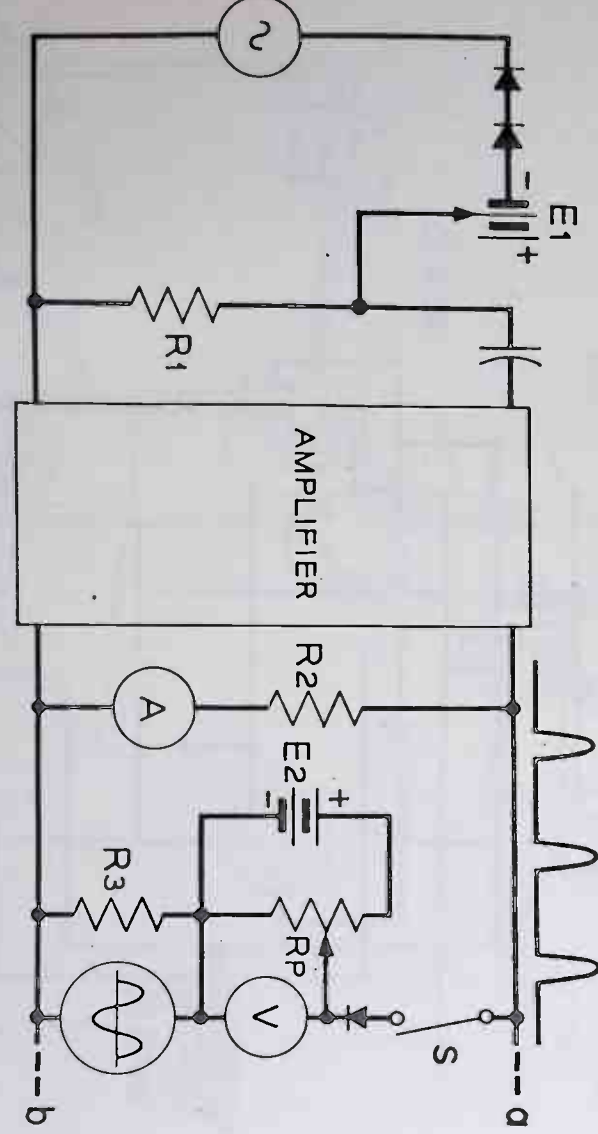


FIGURE 12

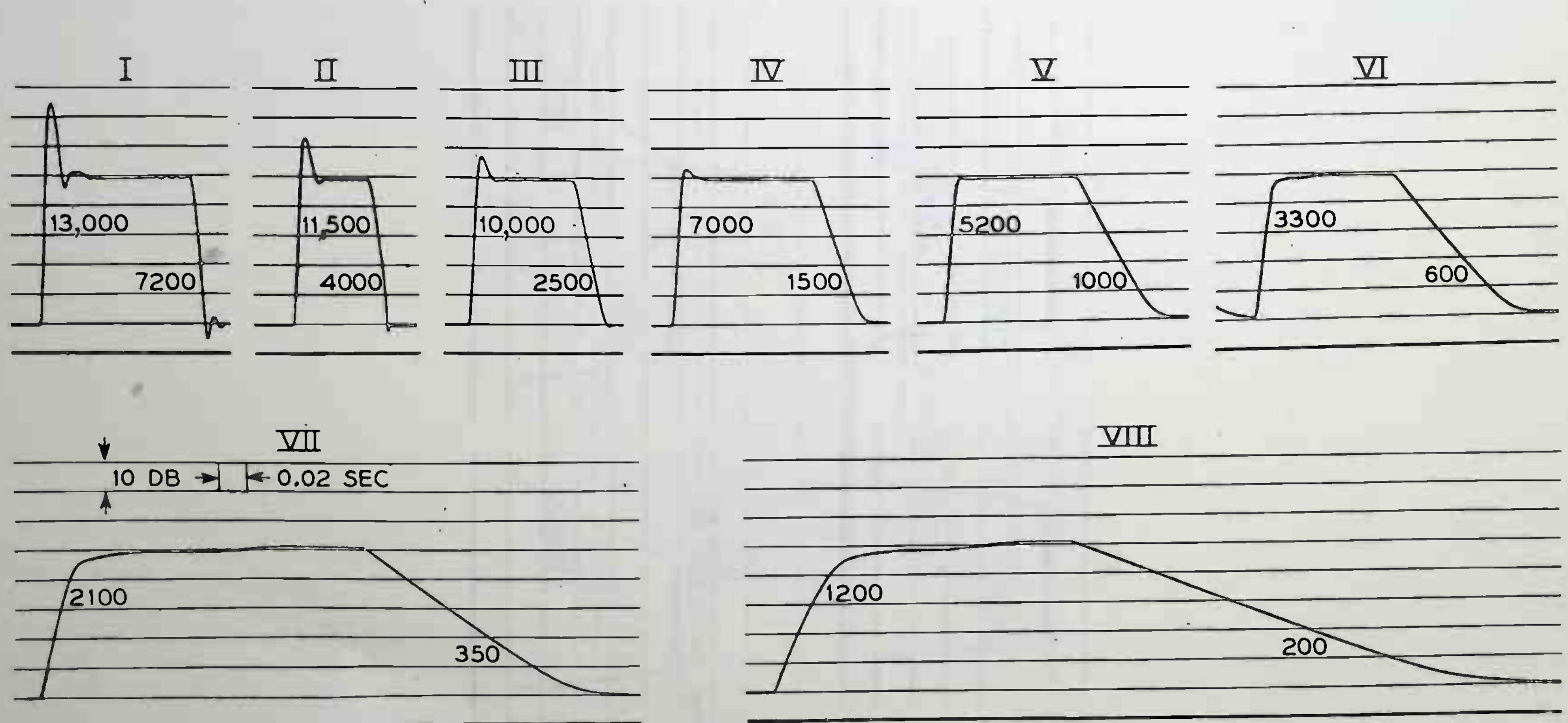


FIGURE 13

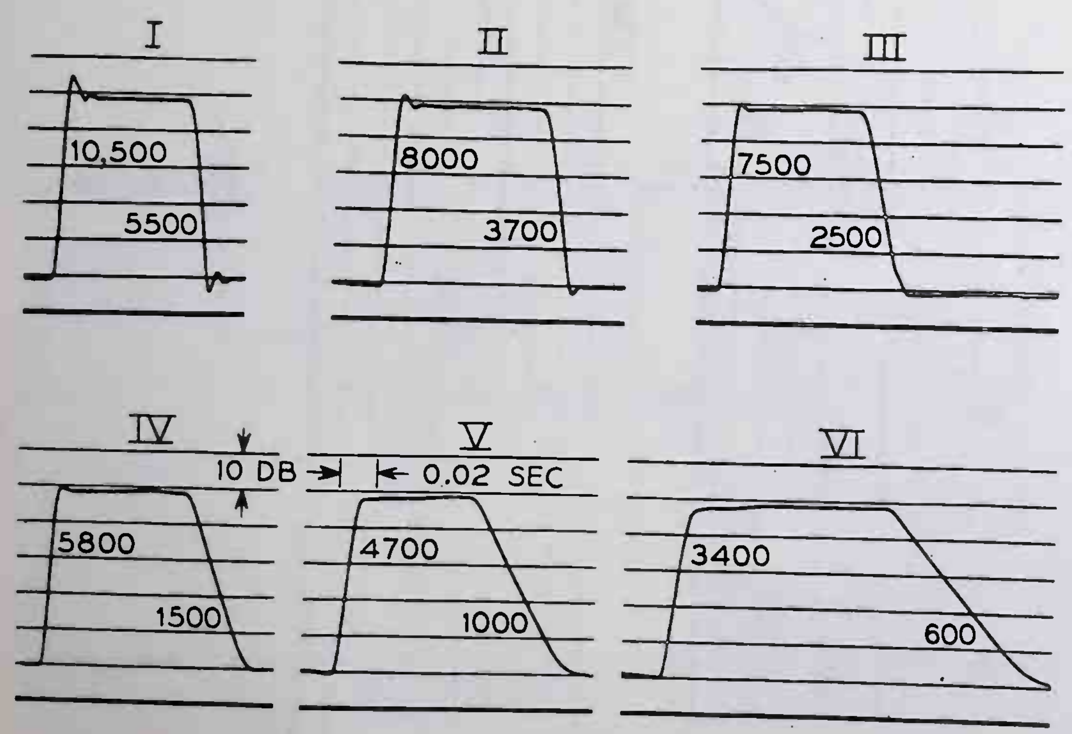


FIGURE 14

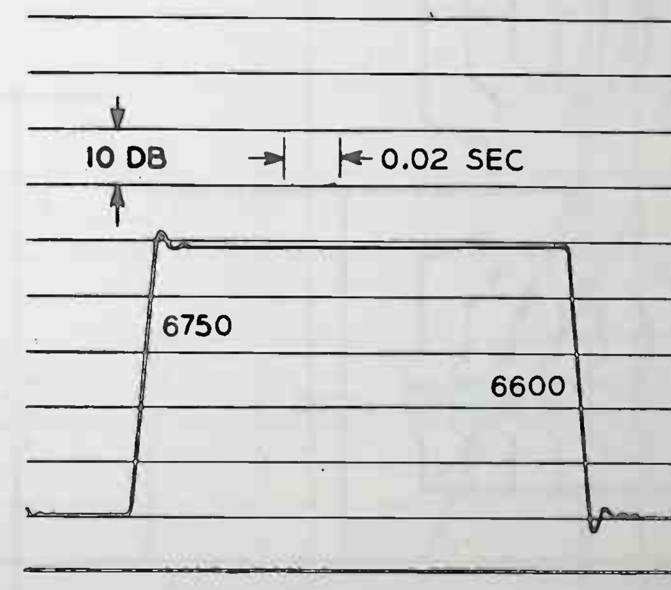


FIGURE 15

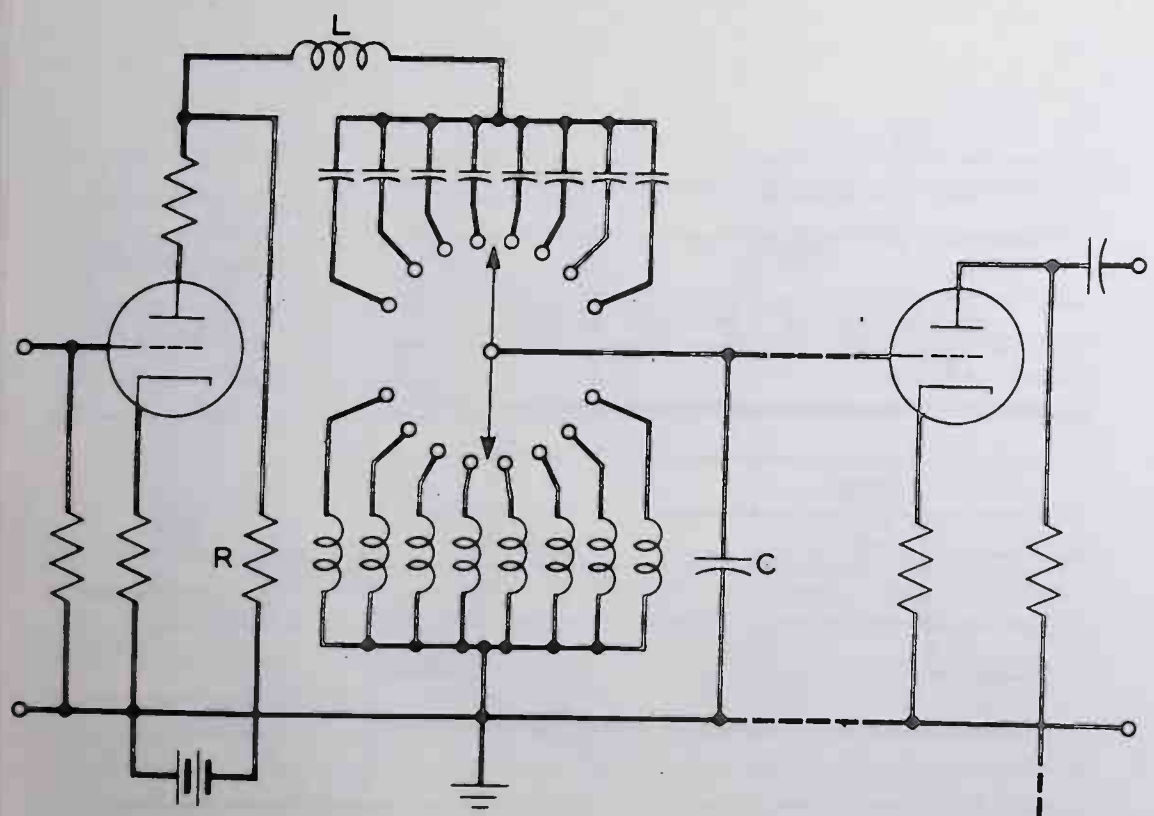


FIGURE 16

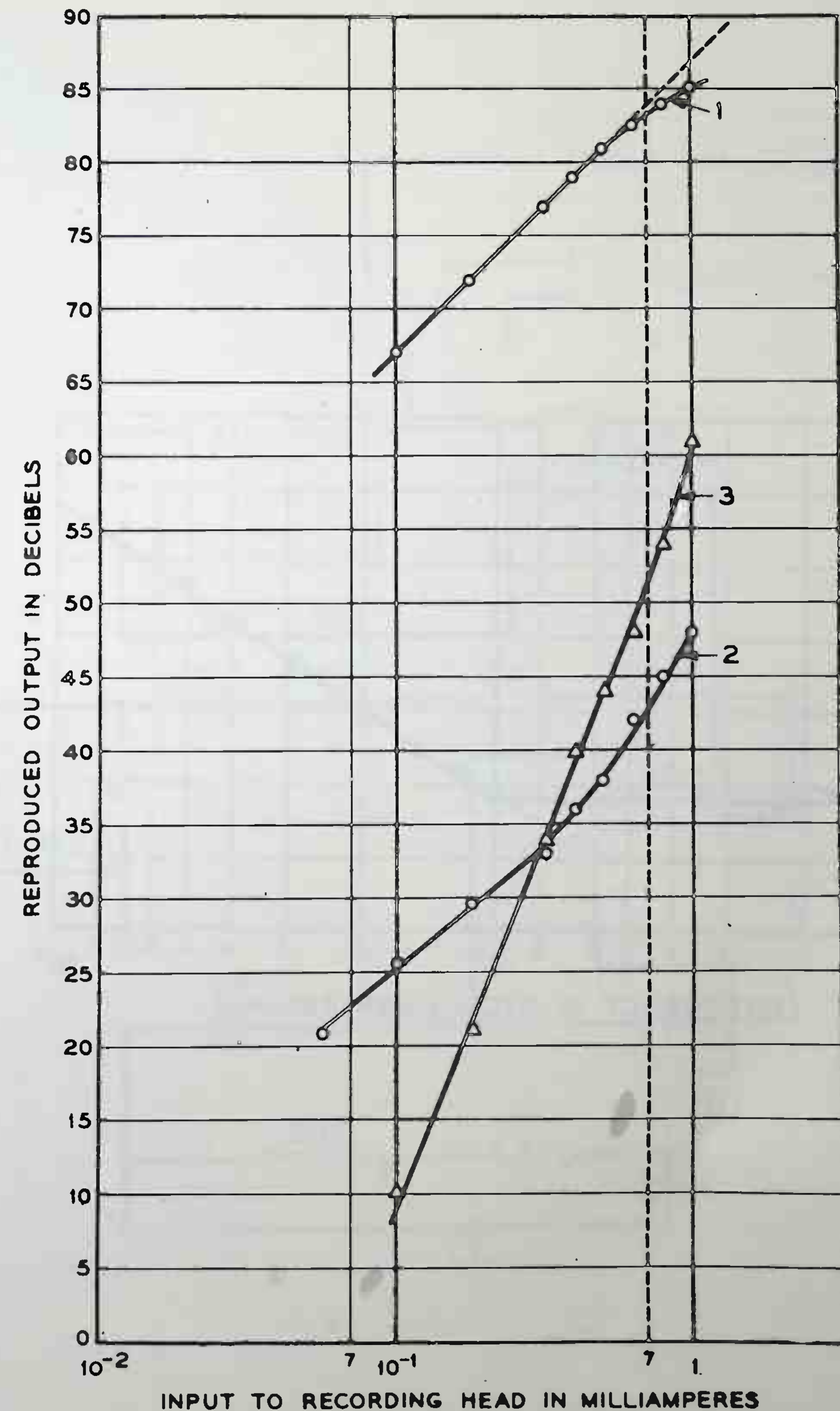


FIGURE 17

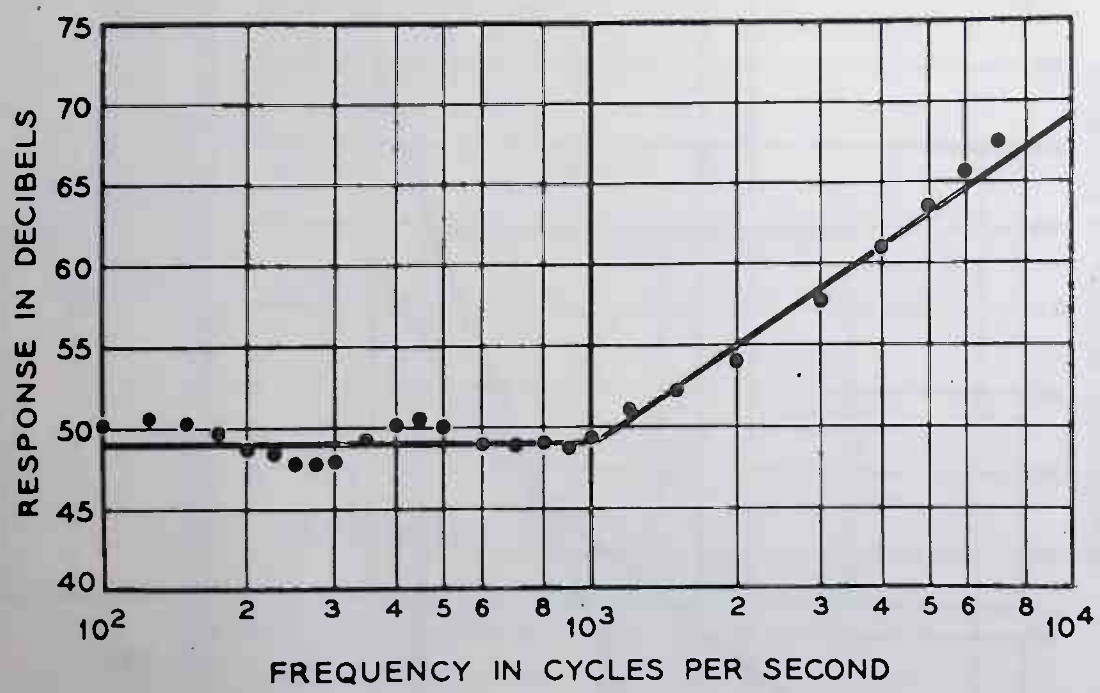


FIGURE 18

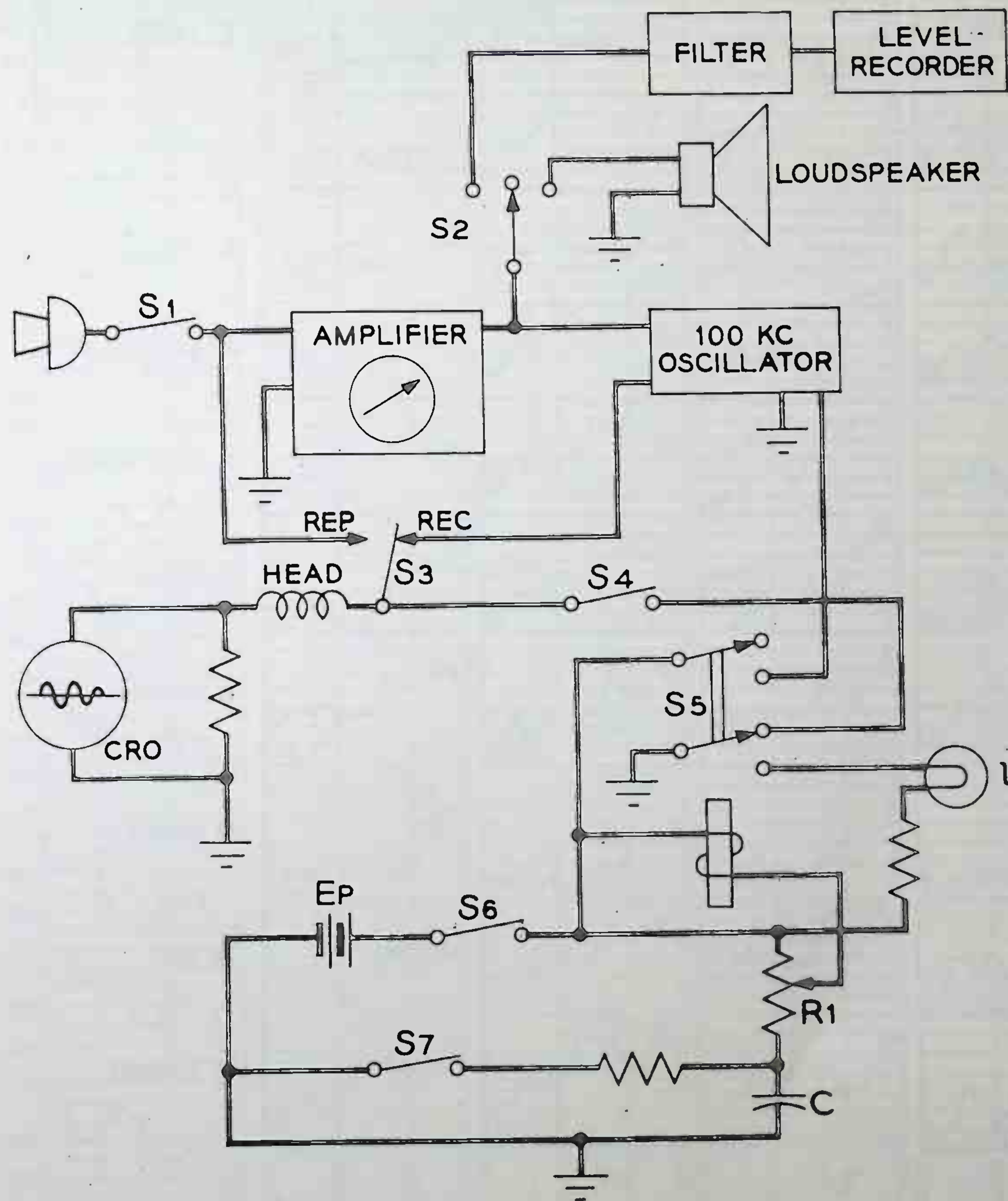
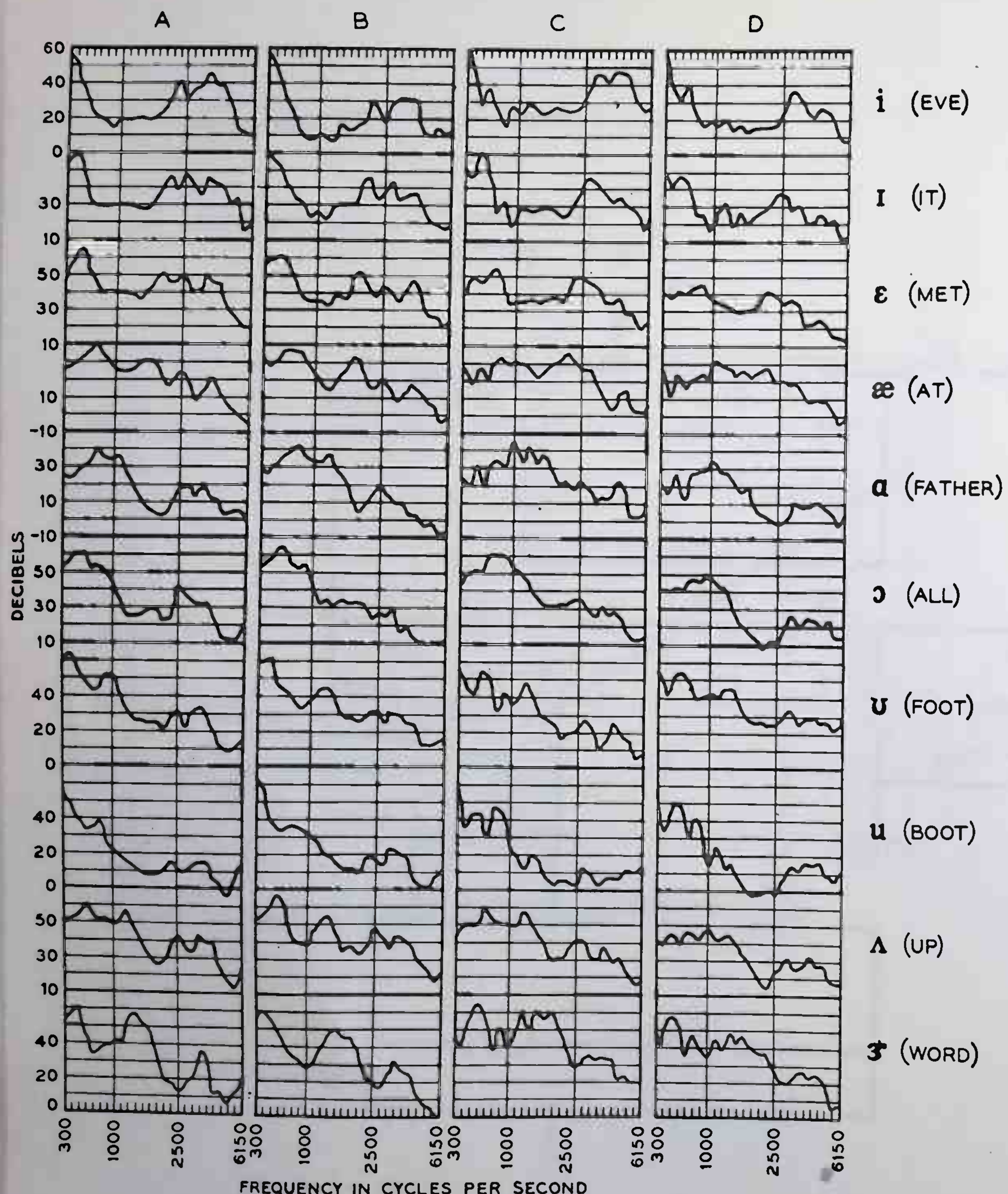


FIGURE 19



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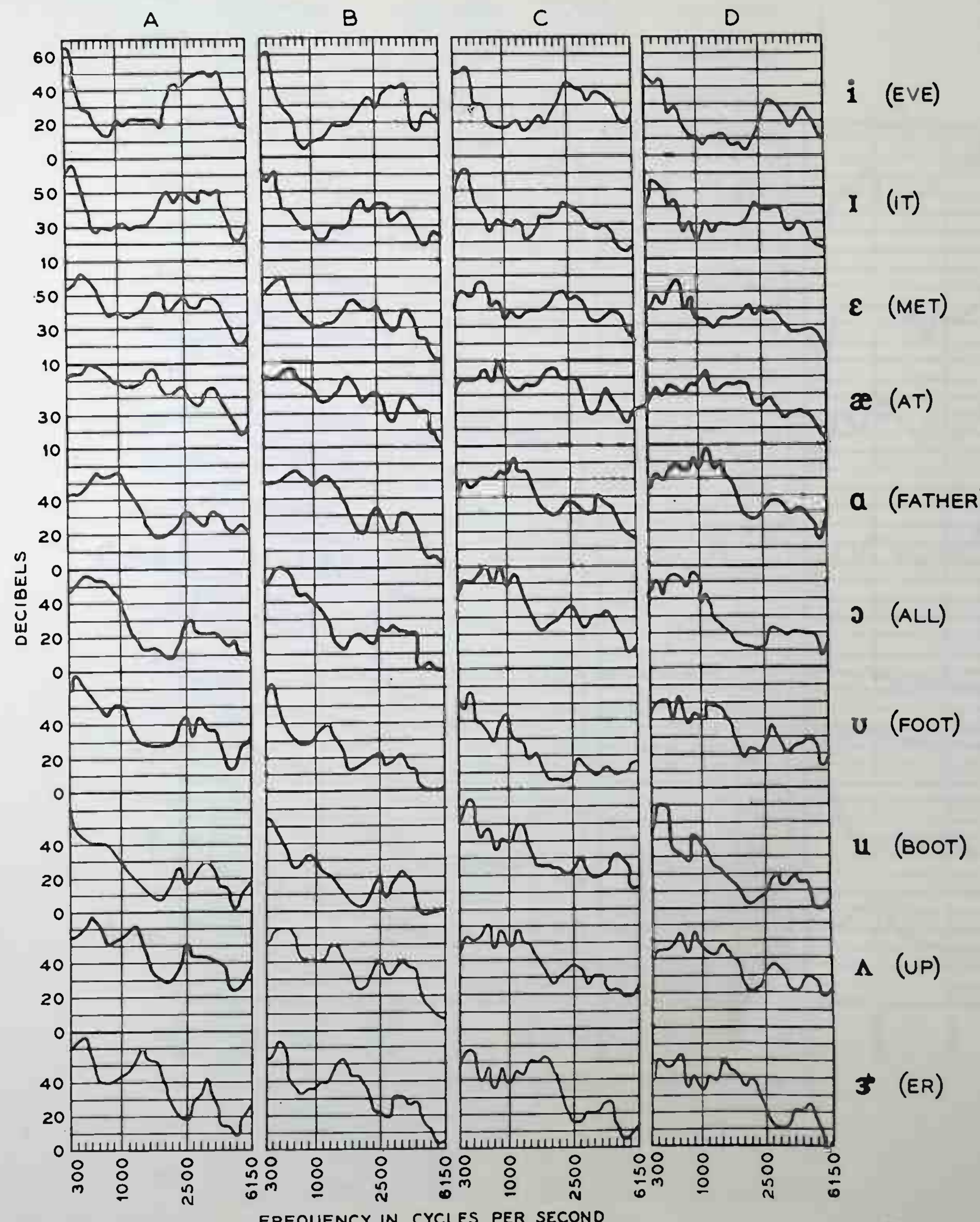

FIGURE 21

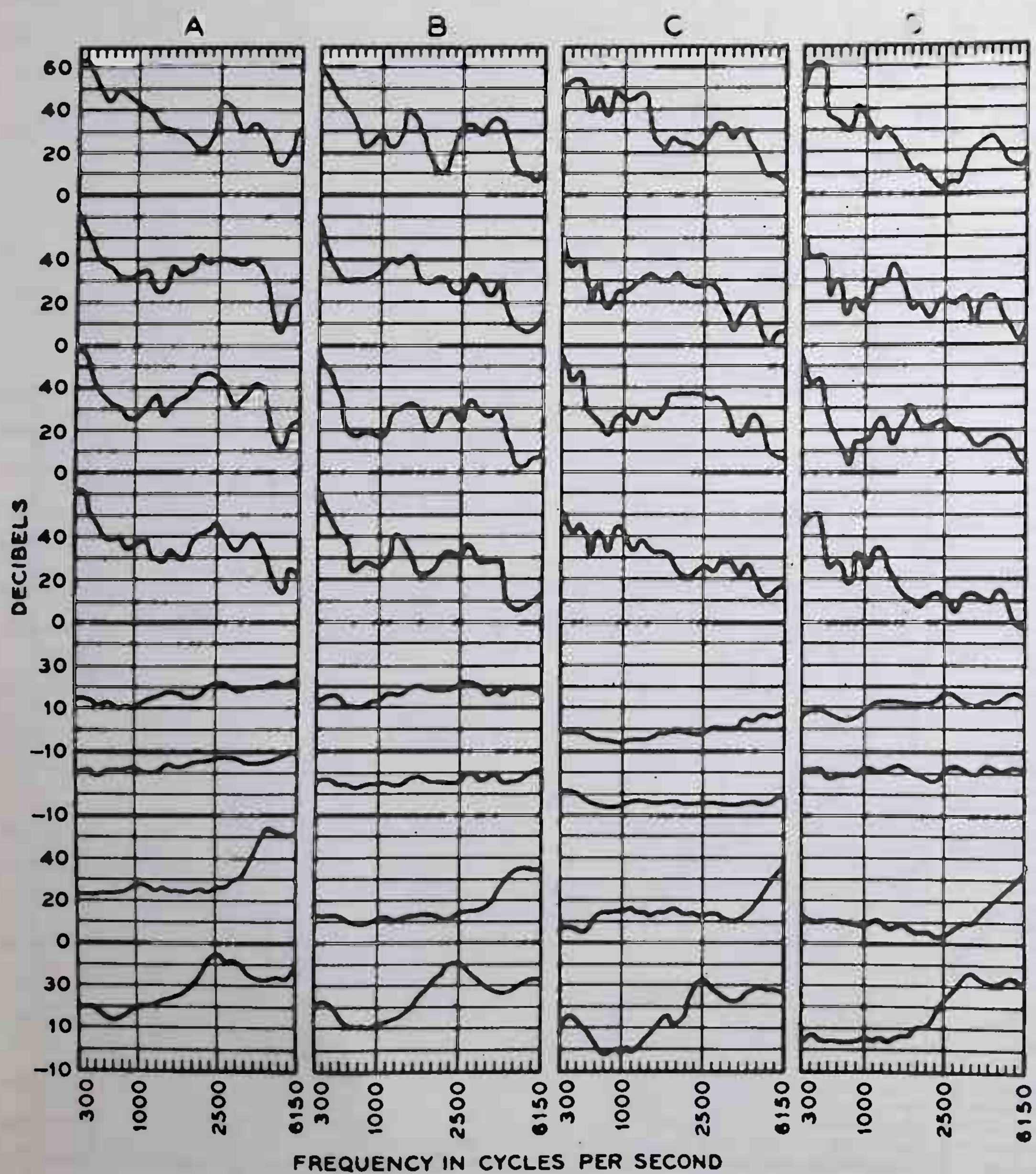
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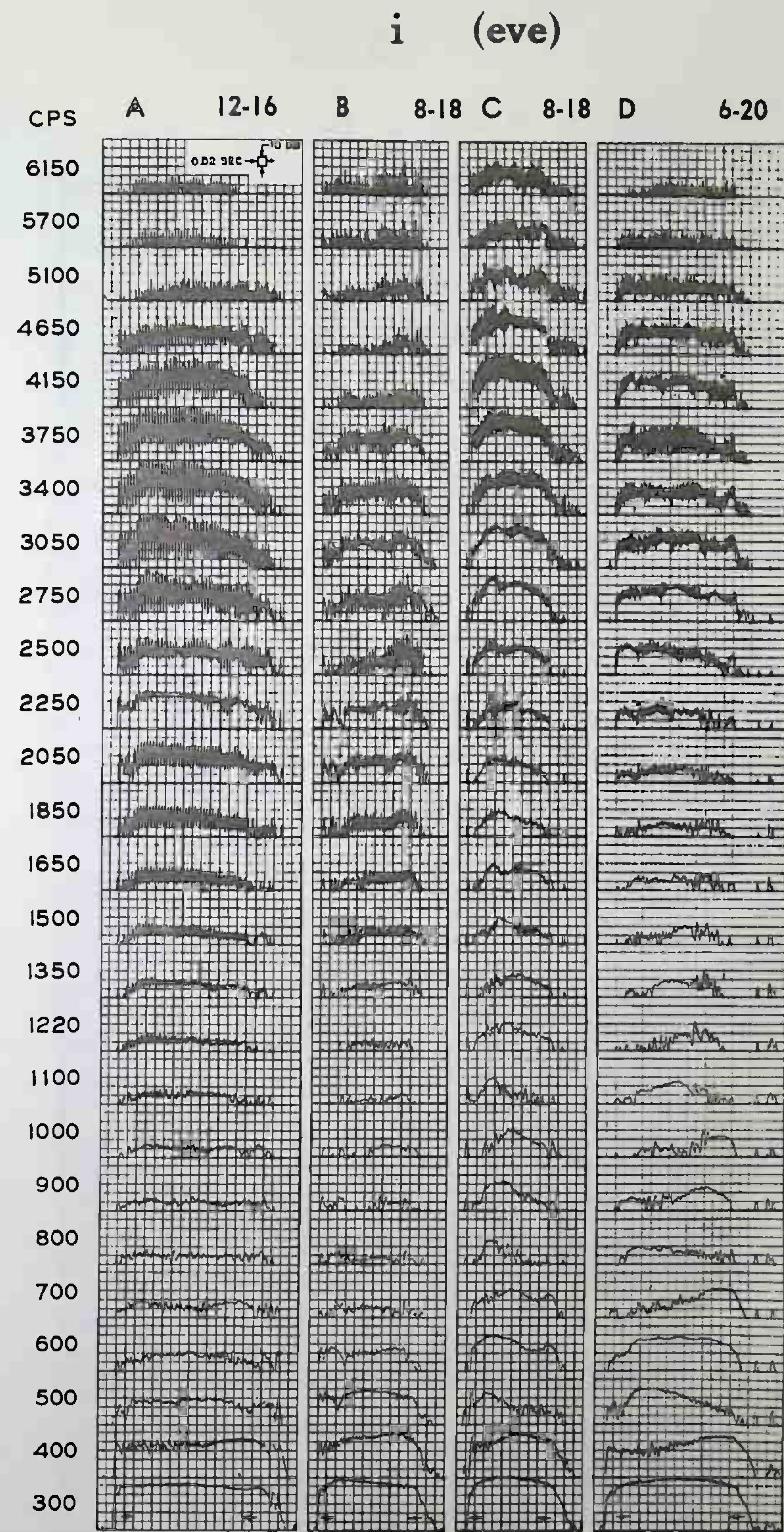
FIGURE 22


i (eve)
CPS
A 12-16
B 8-18
C 8-18
D 6-20
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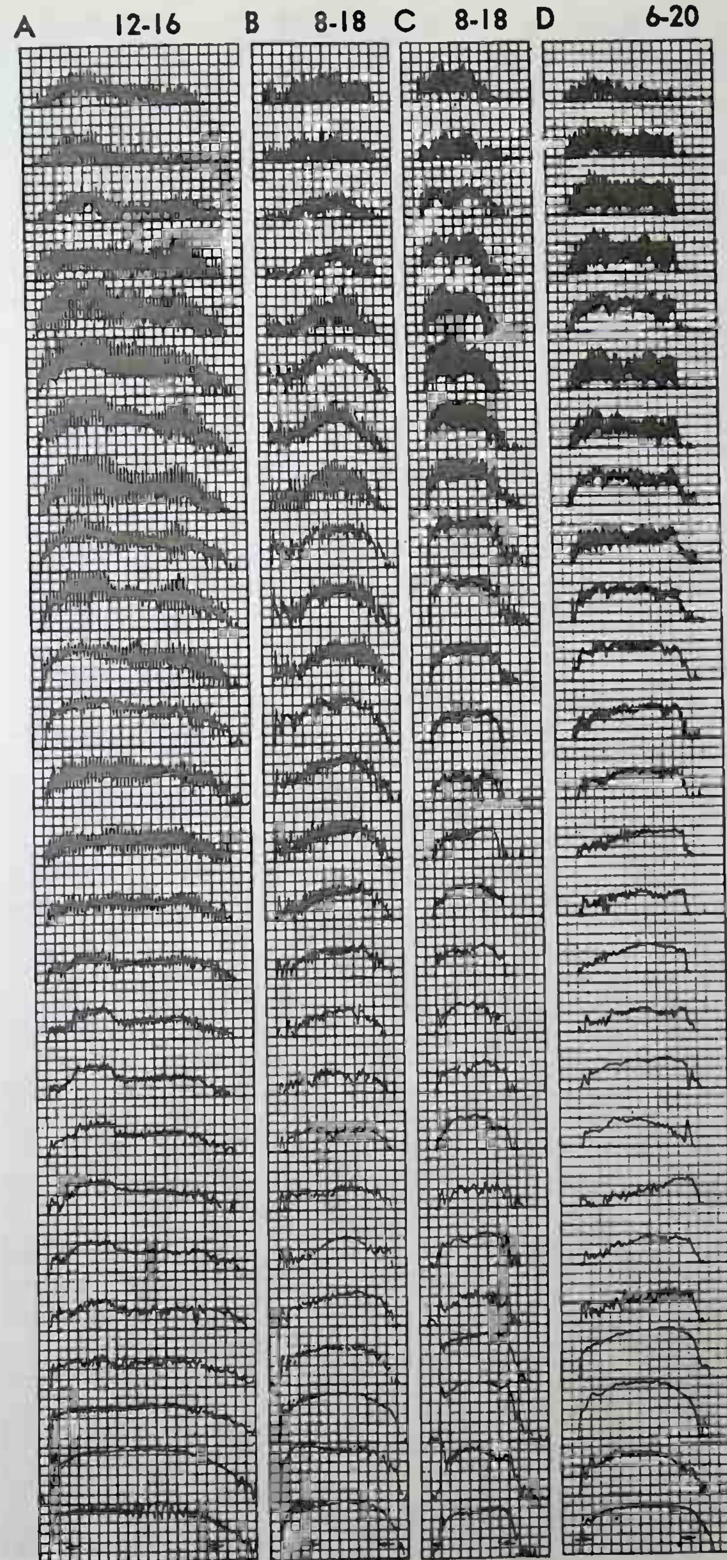
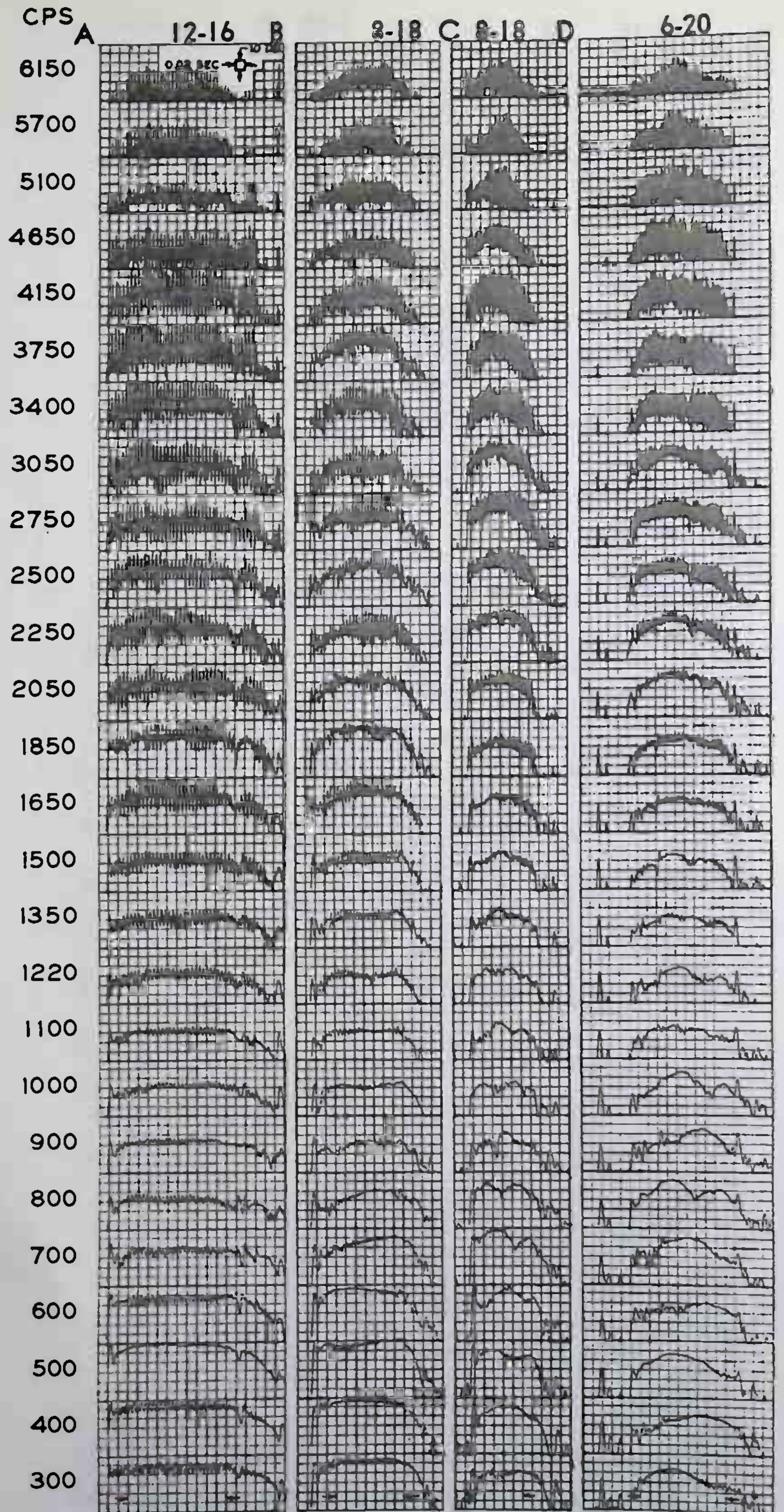
I (it)



ε (met)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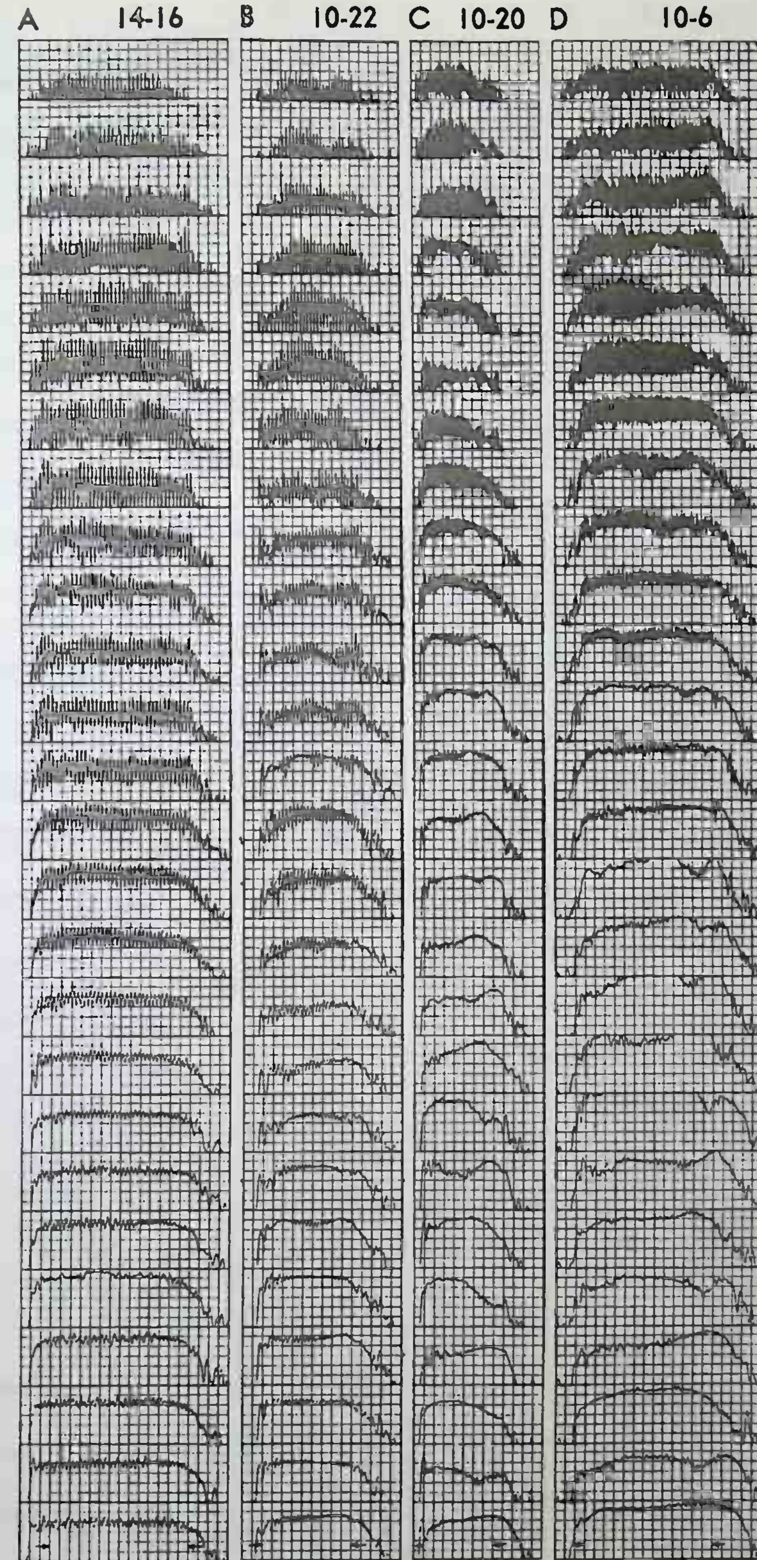
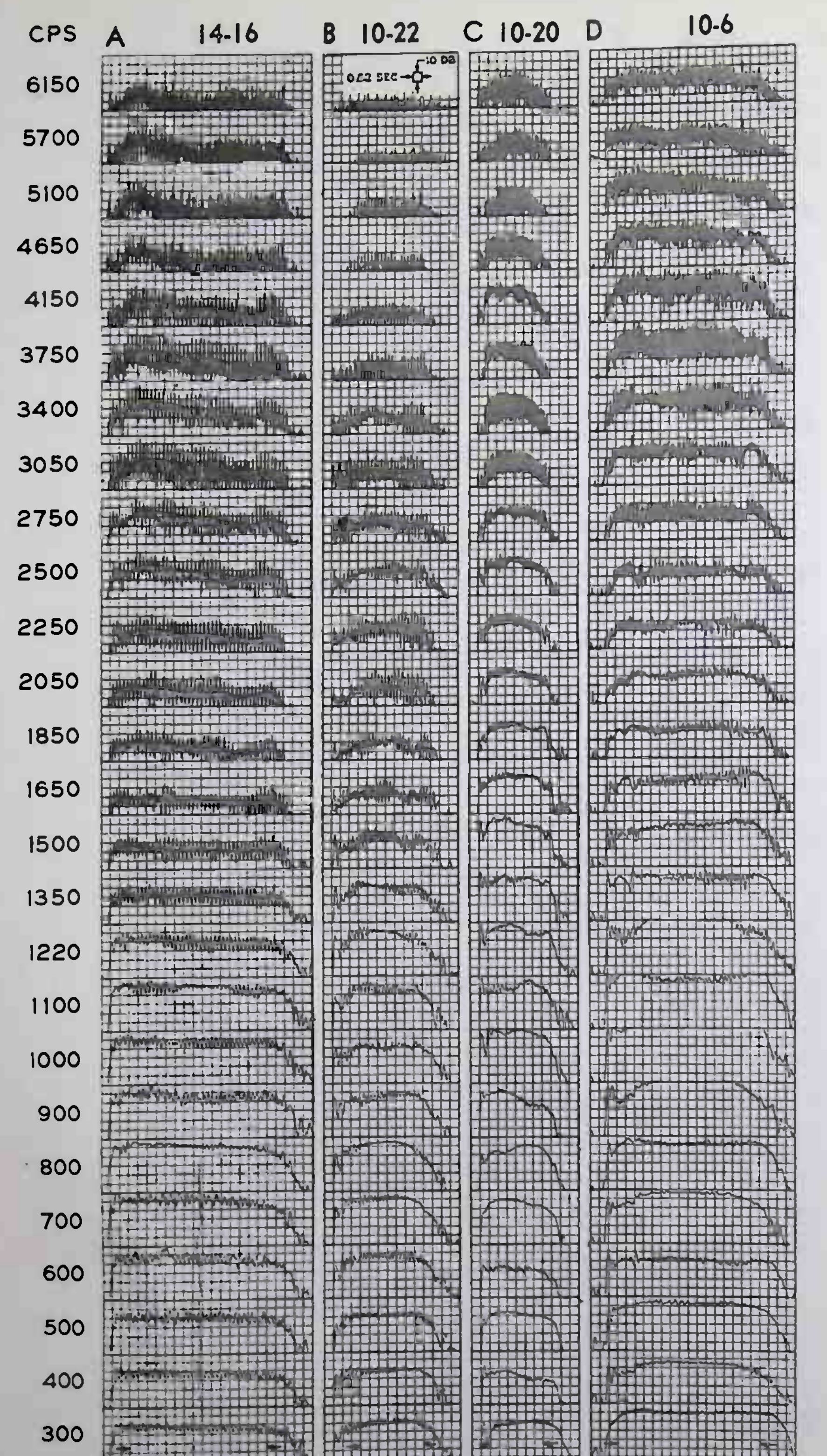
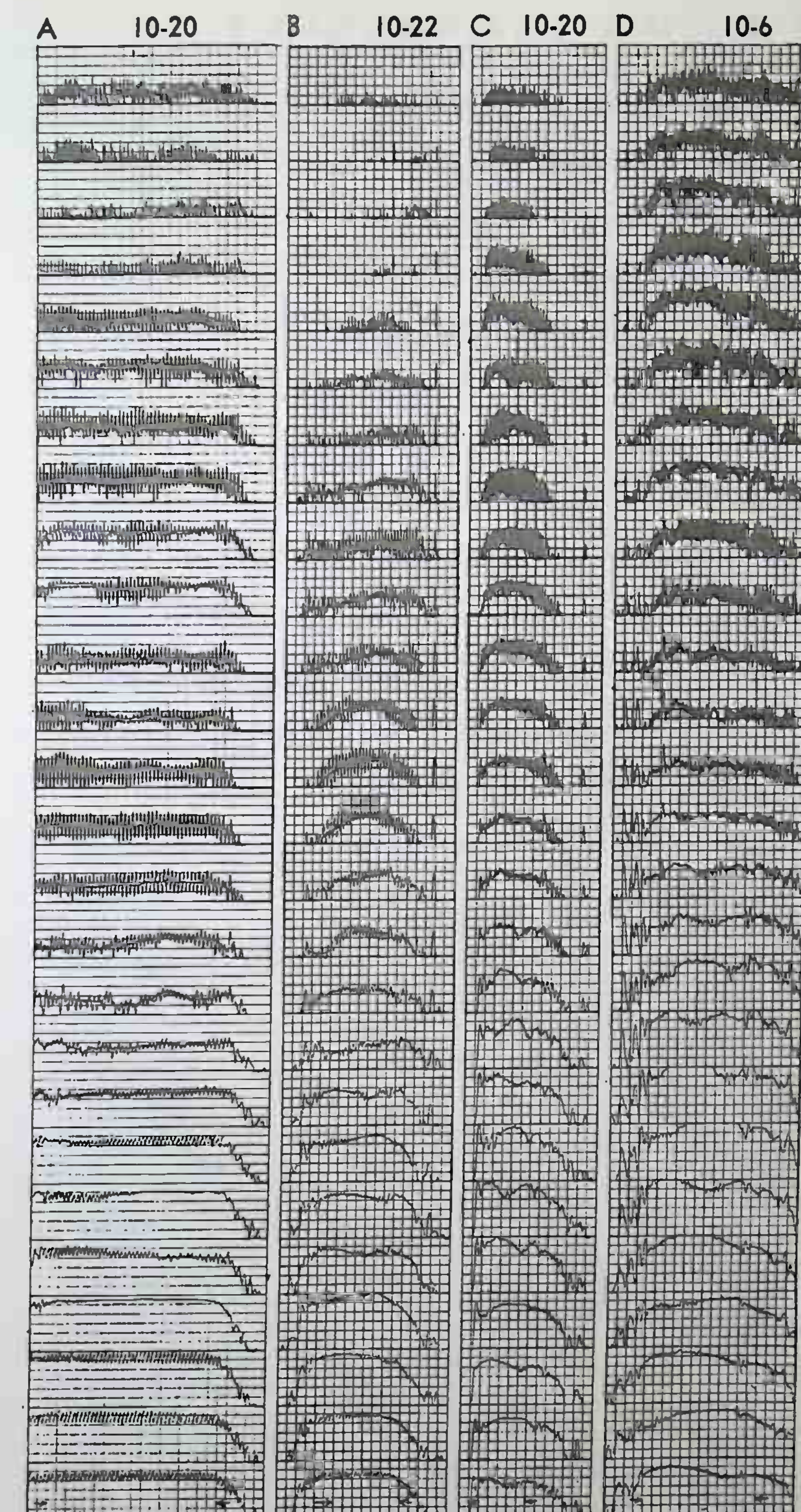
æ (at)


A 14-16
B 10-22
C 10-20
D 10-6

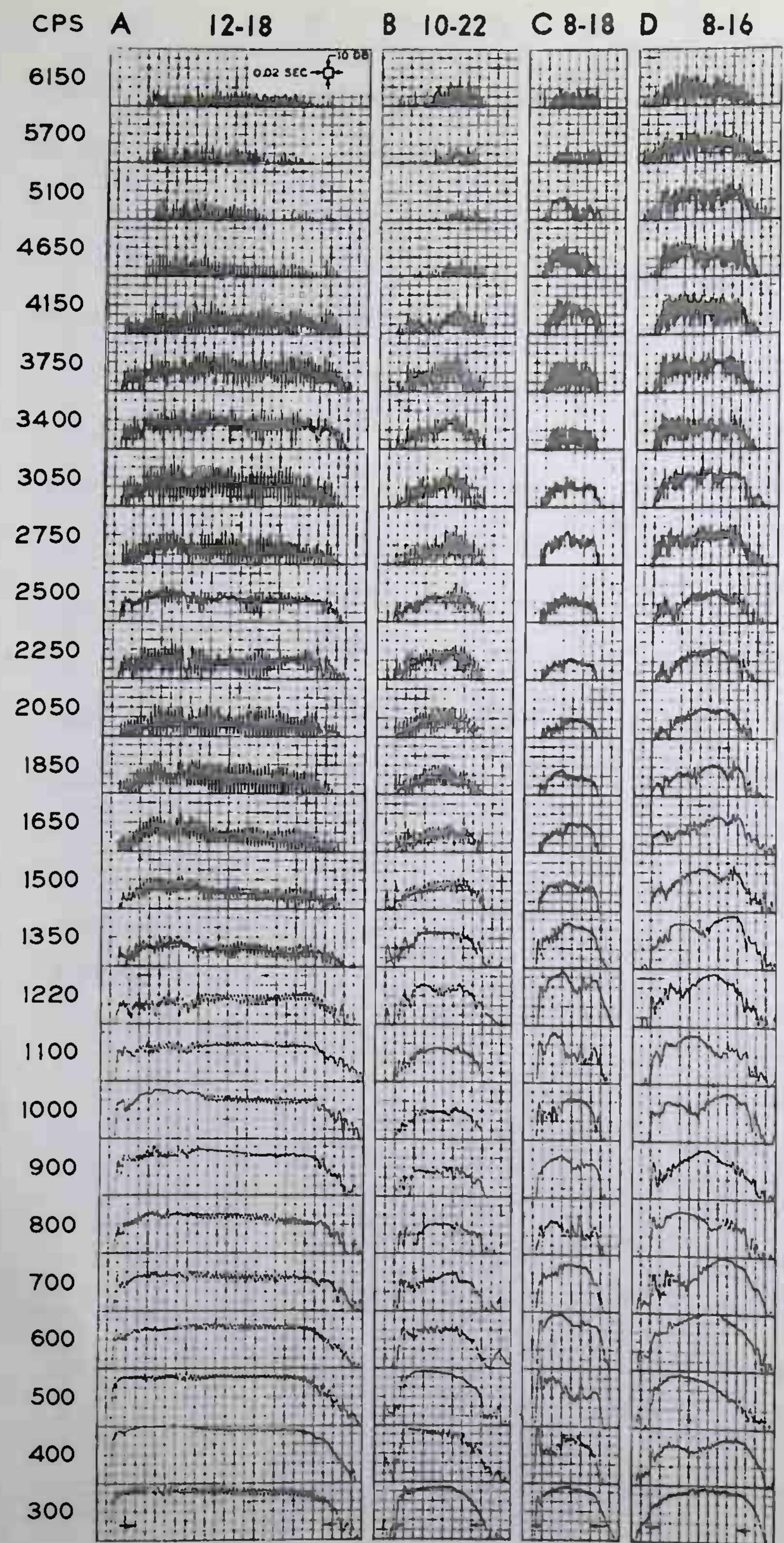
ɑ (father)



ɔ (all)



ʊ (foot)


CPS
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B 10-22
C 8-18
D 8-16
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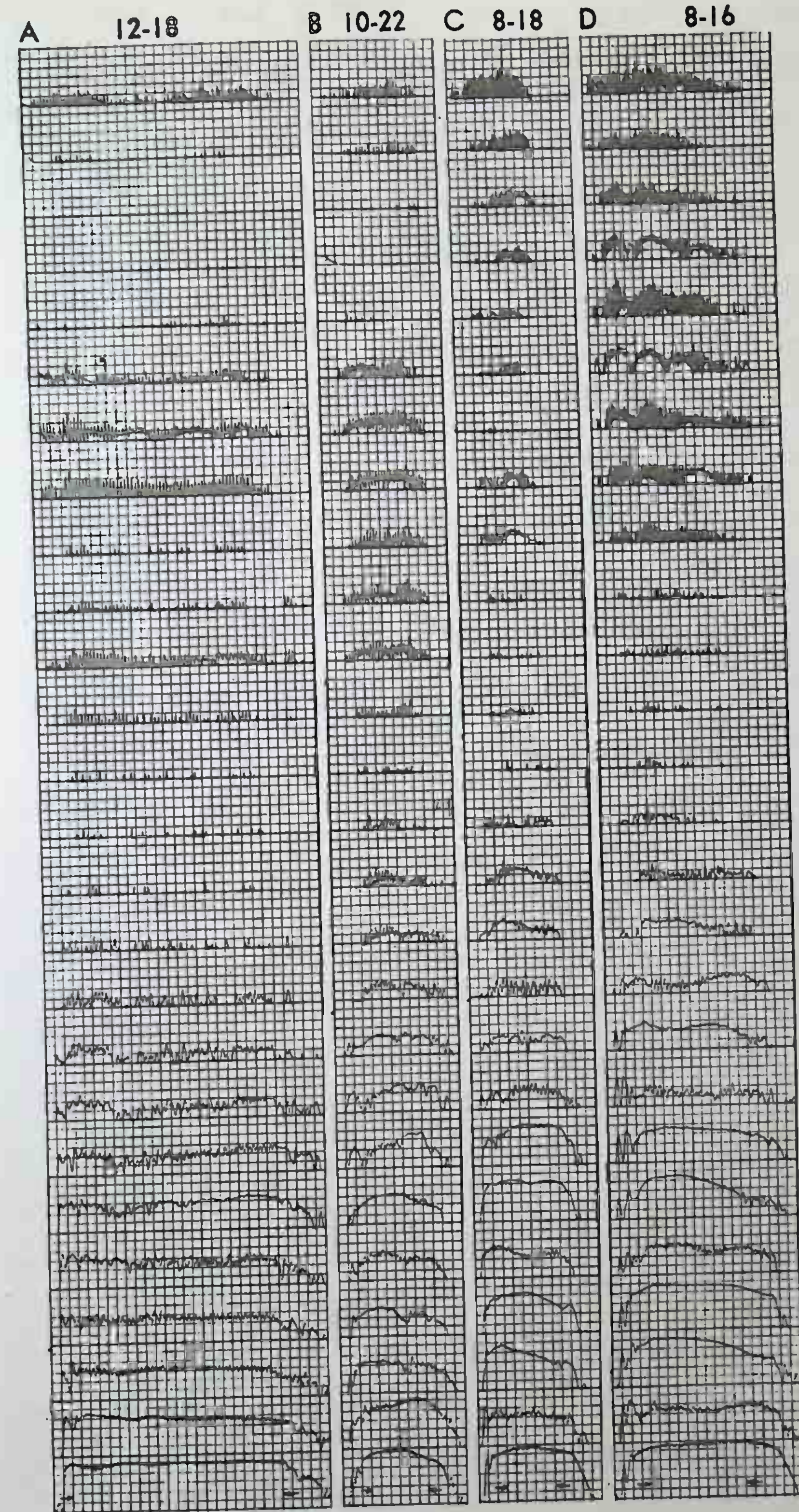
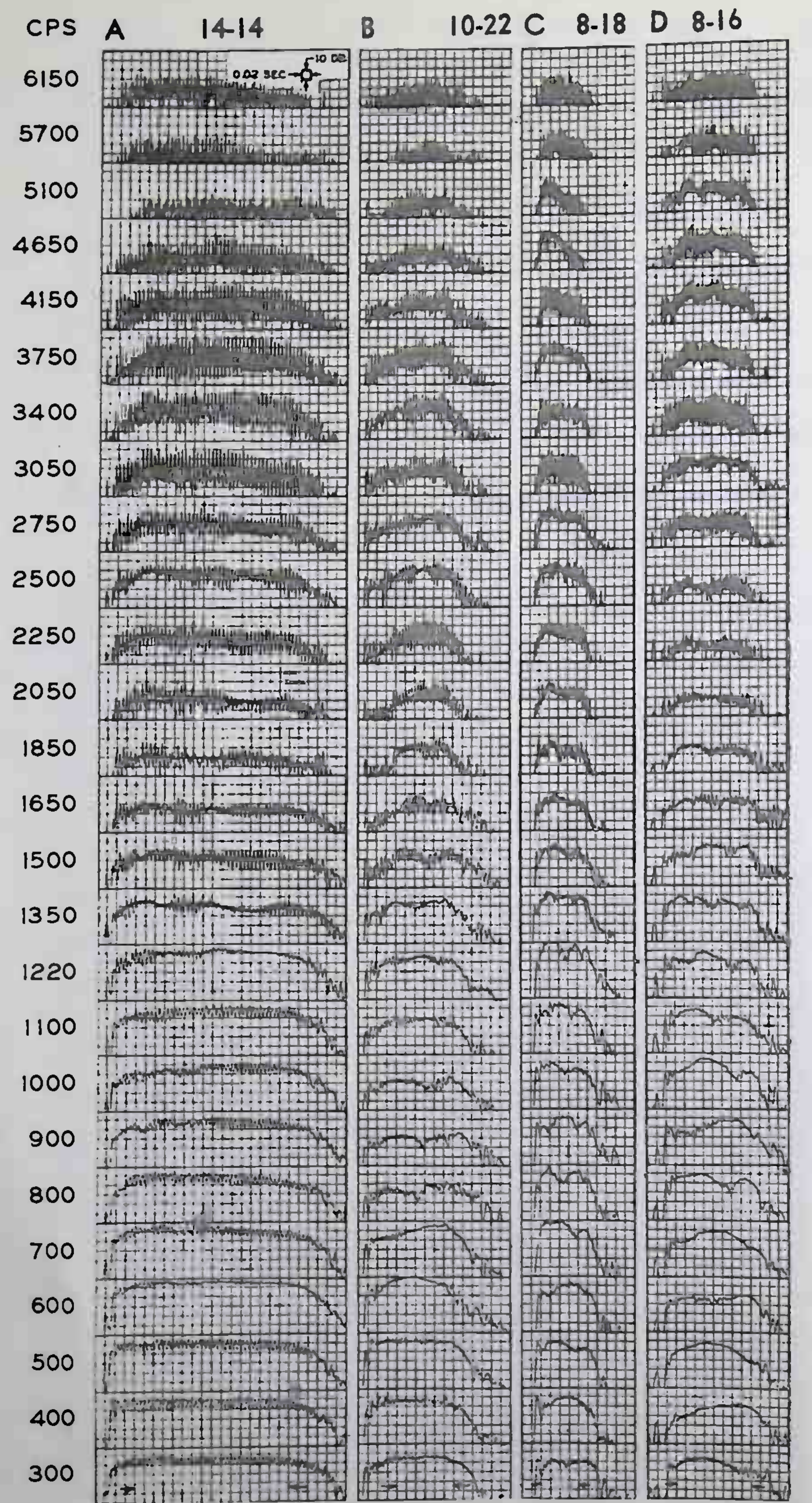
u (boot)


A 12-18
B 10-22
C 8-18
D 8-16

ʌ (up)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B 10-22
C 8-18
D 8-16
0.02 SEC
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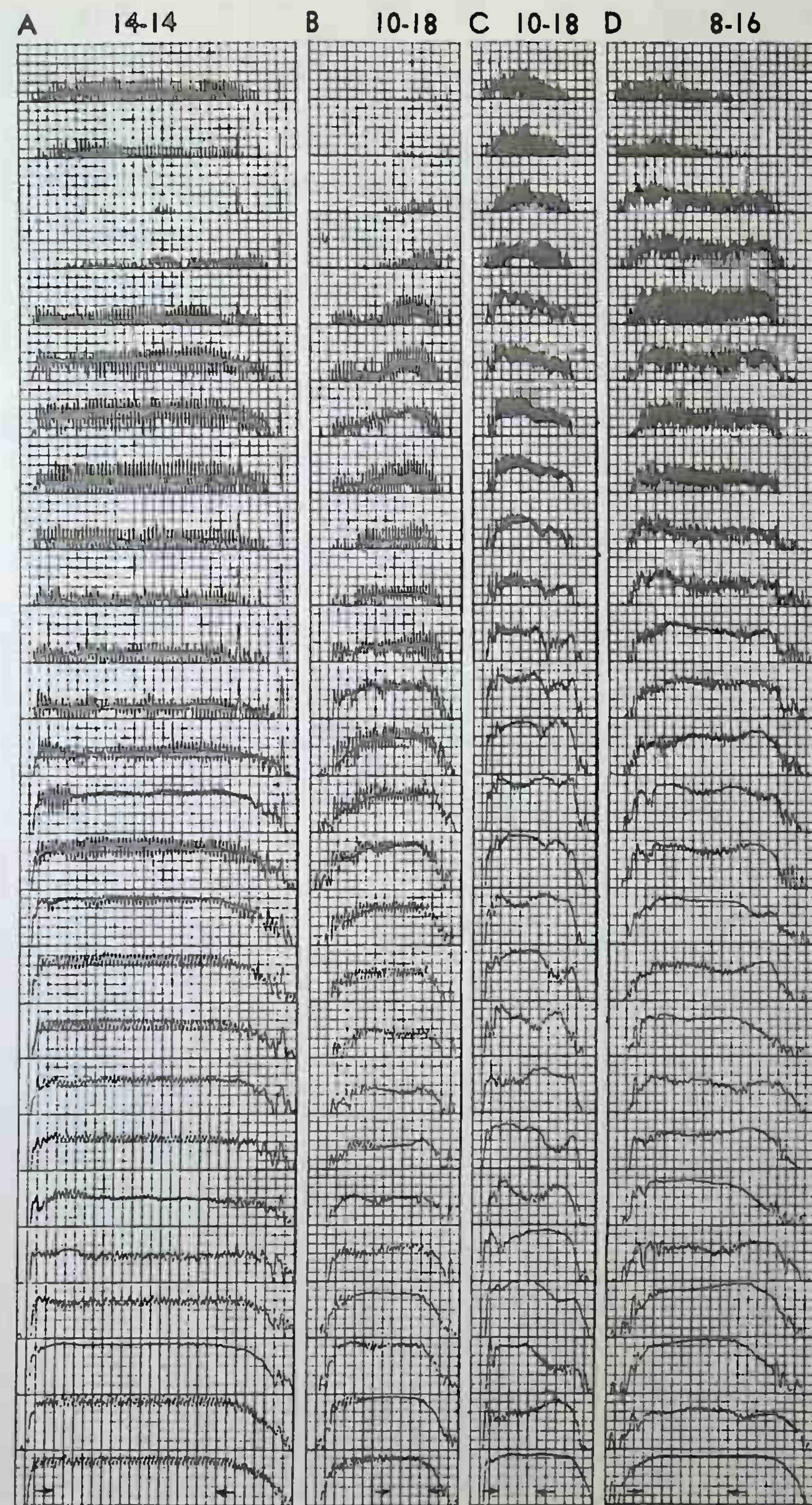
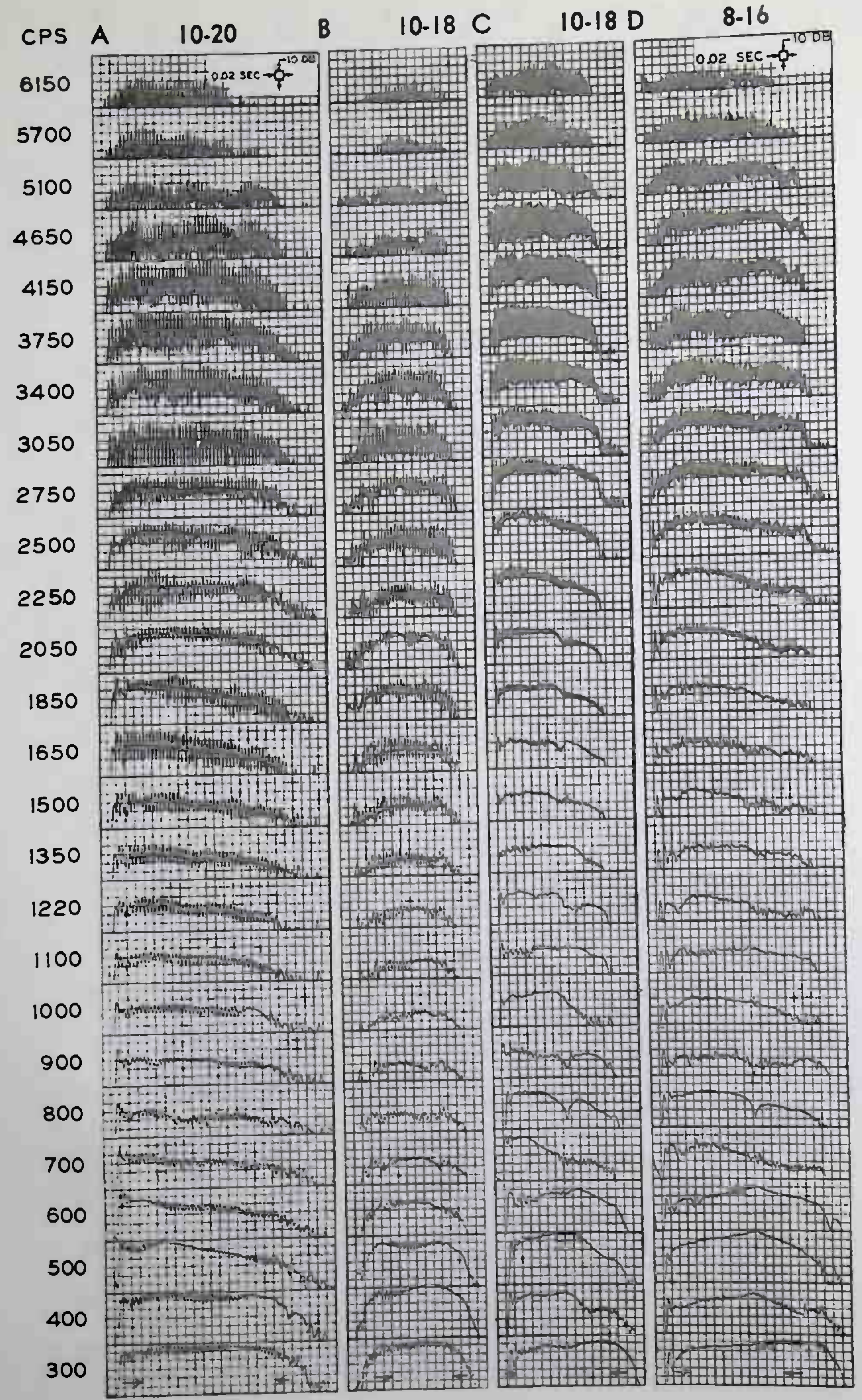
ɜ (word)


A 14-14
B 10-18
C 10-18
D 8-16

eɪ (say)


CPS
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D 8-16
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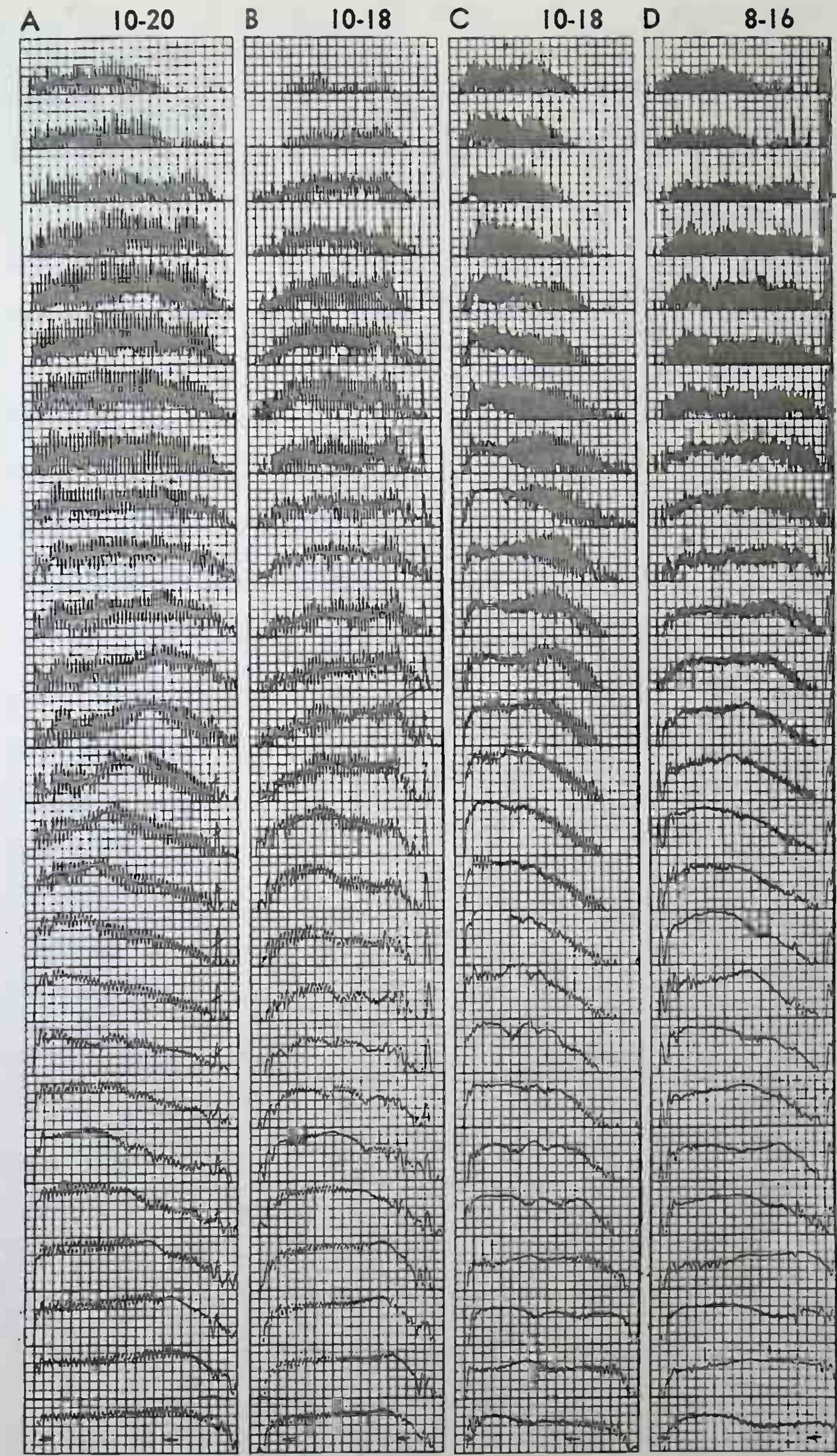
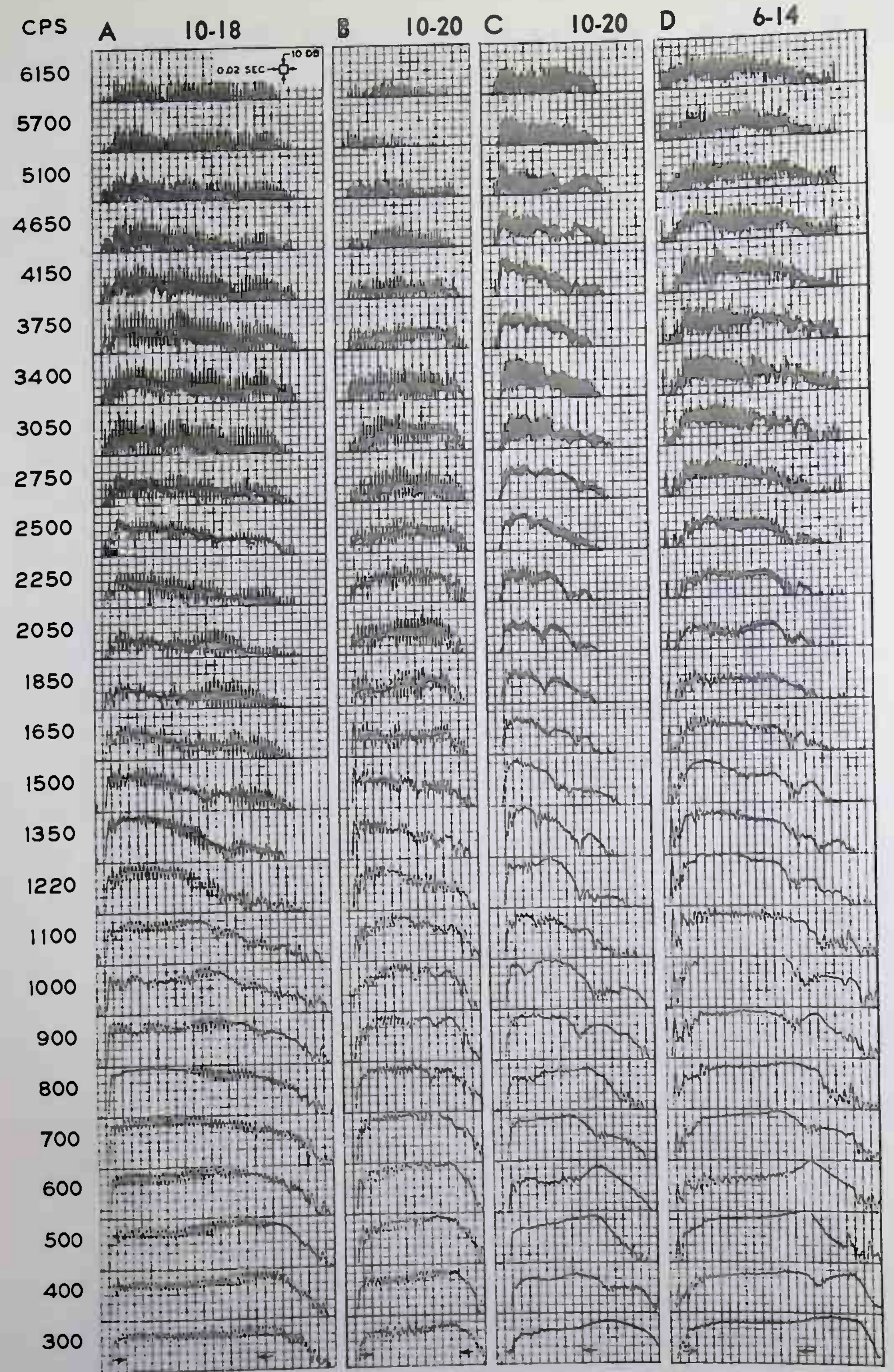
aɪ (I)


A 10-20
B 10-18
C 10-18
D 8-16

aʊ (out)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B 10-20
C 10-20
D 6-14
0.02 SEC
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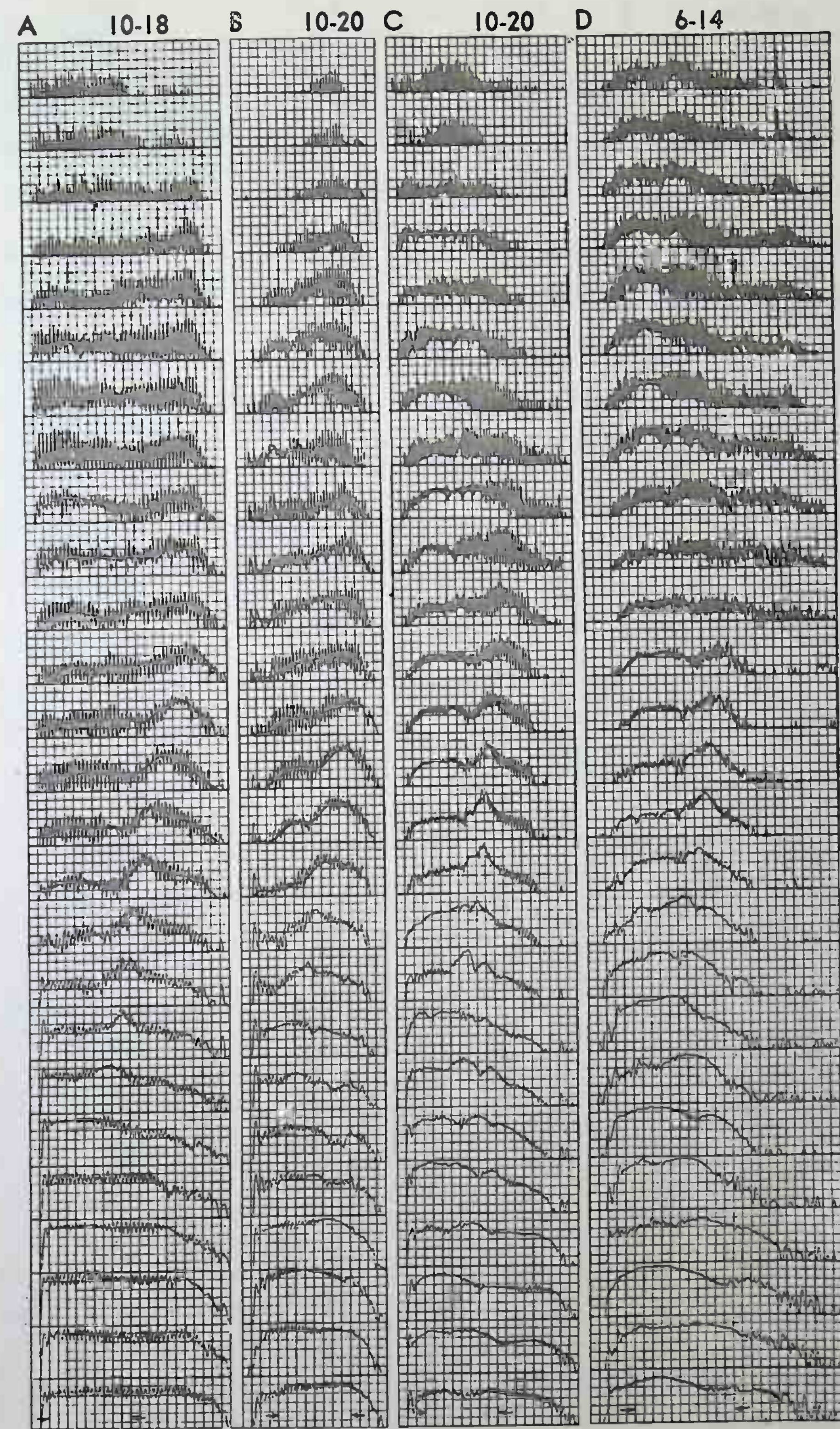
ɔɪ (boy)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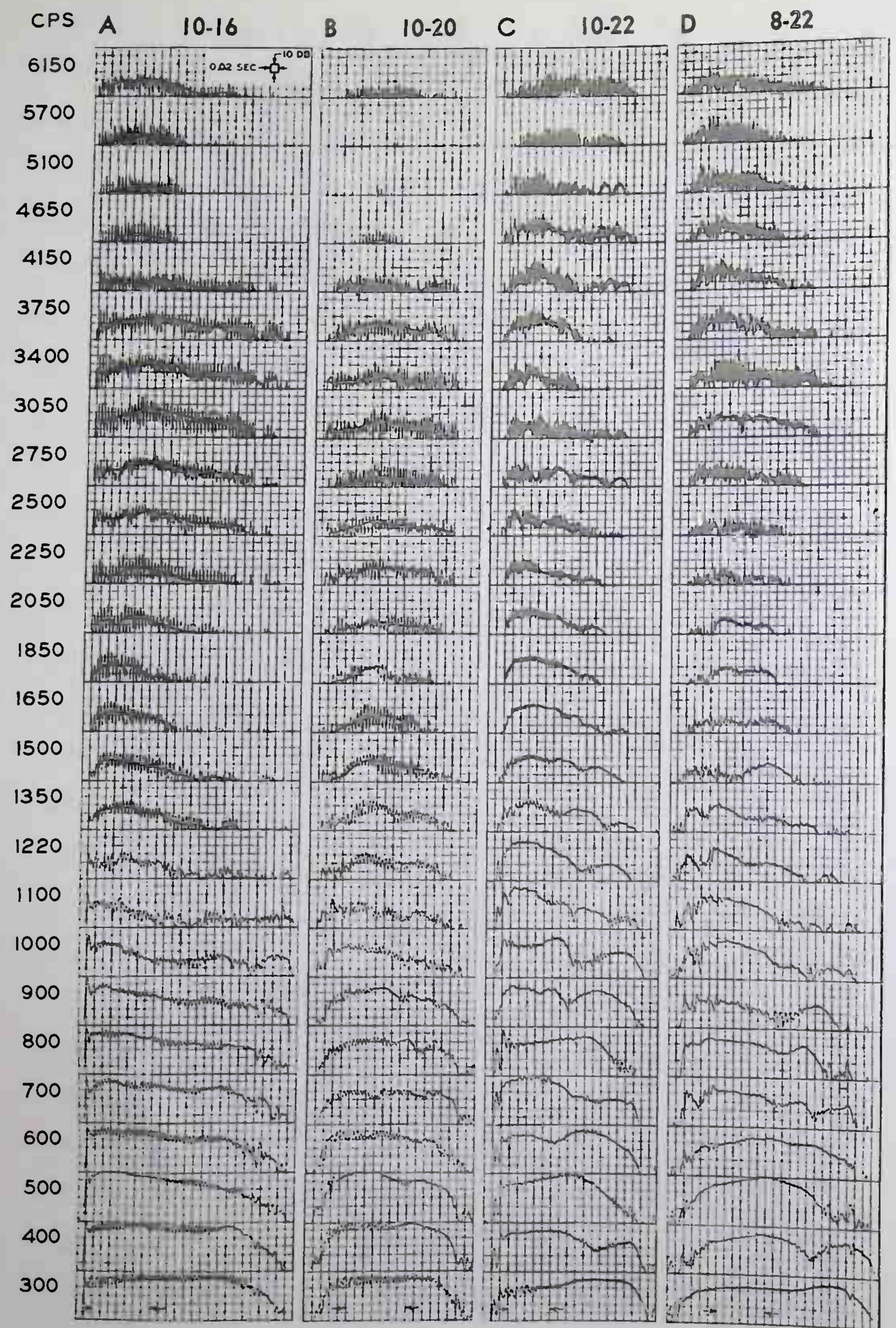
B 10-20
C 10-20
D 6-14

ou (go)


CPS
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D 8-22
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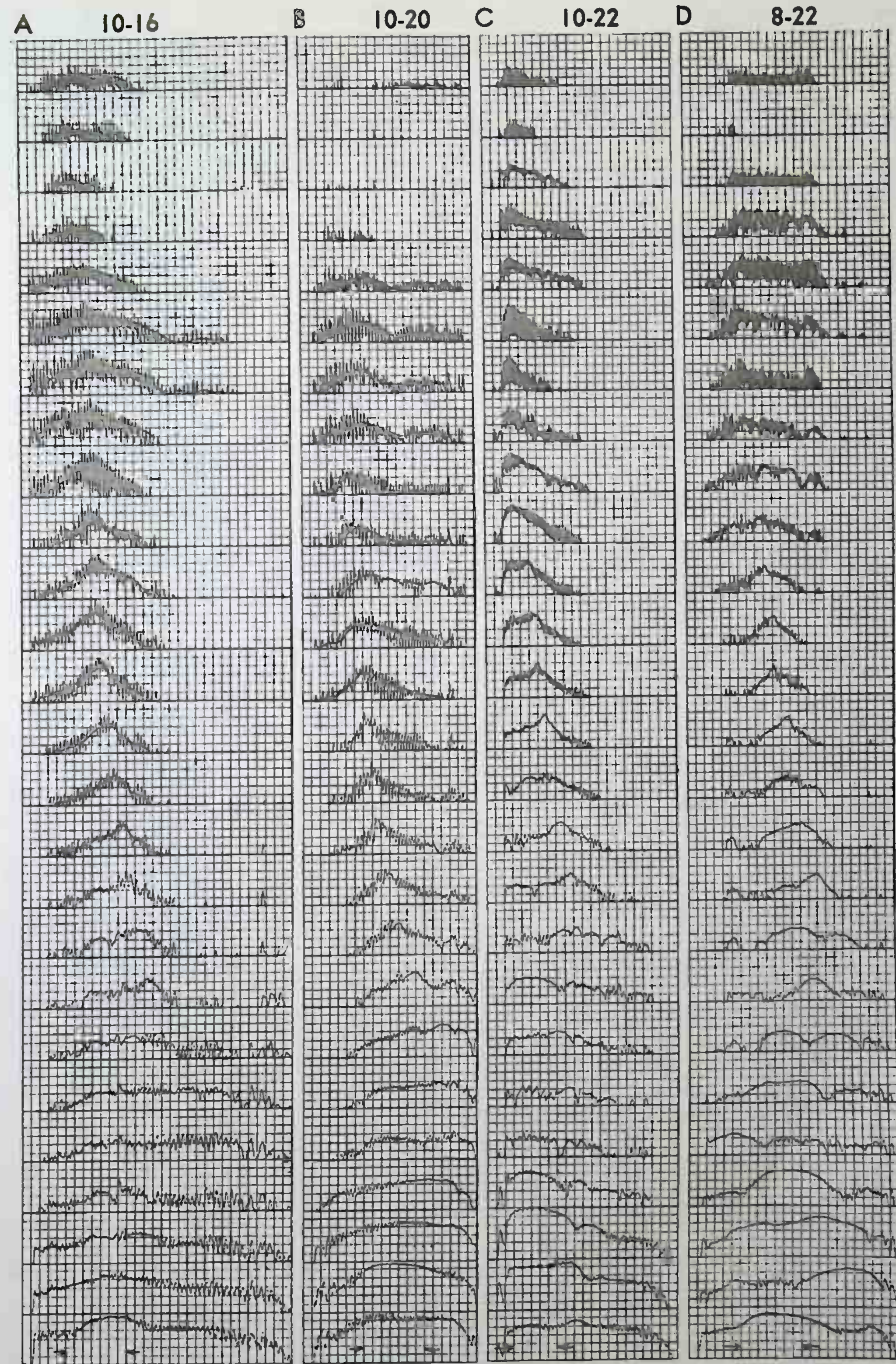
ju (you)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C 10-22
D 8-22

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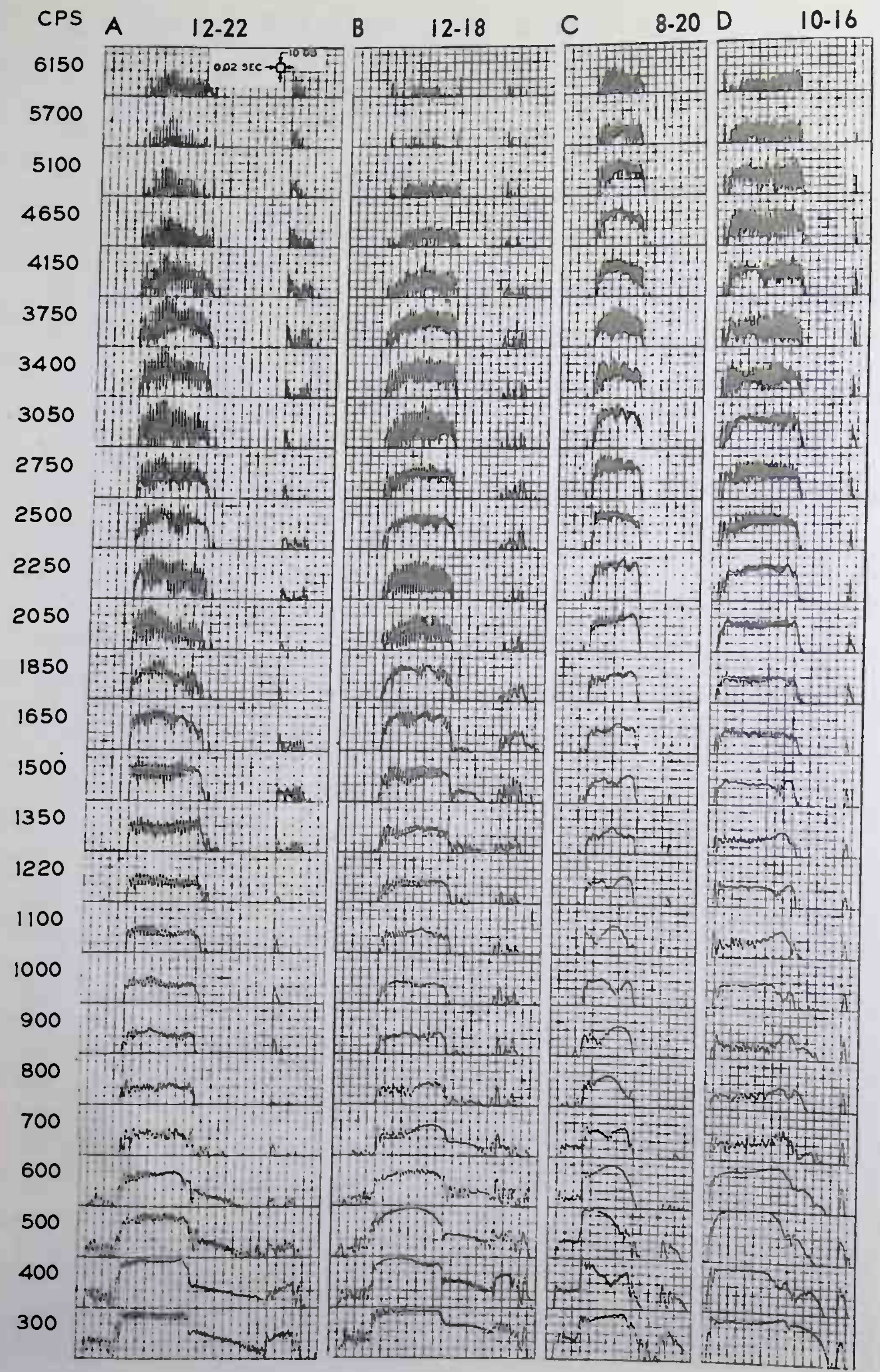
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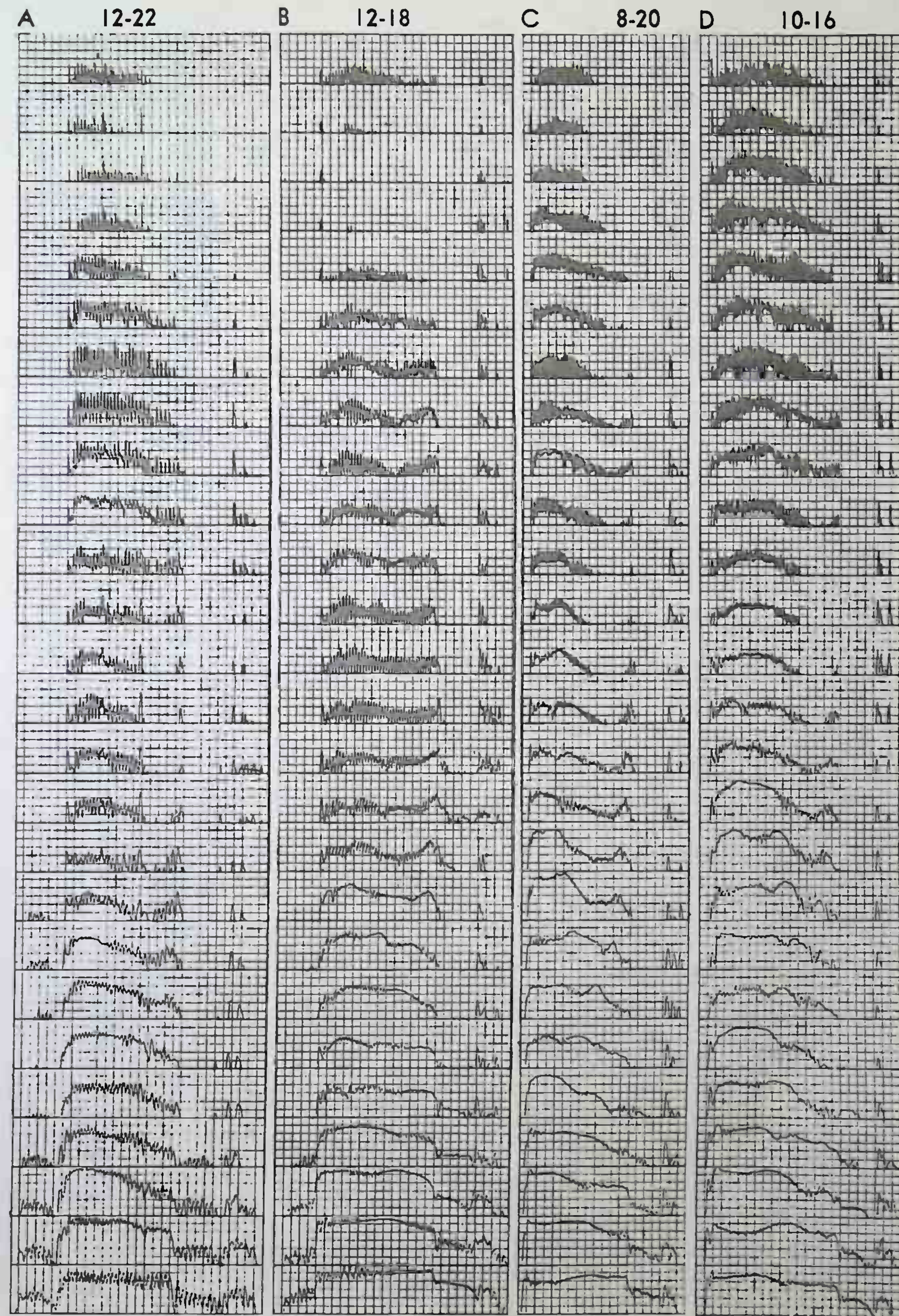
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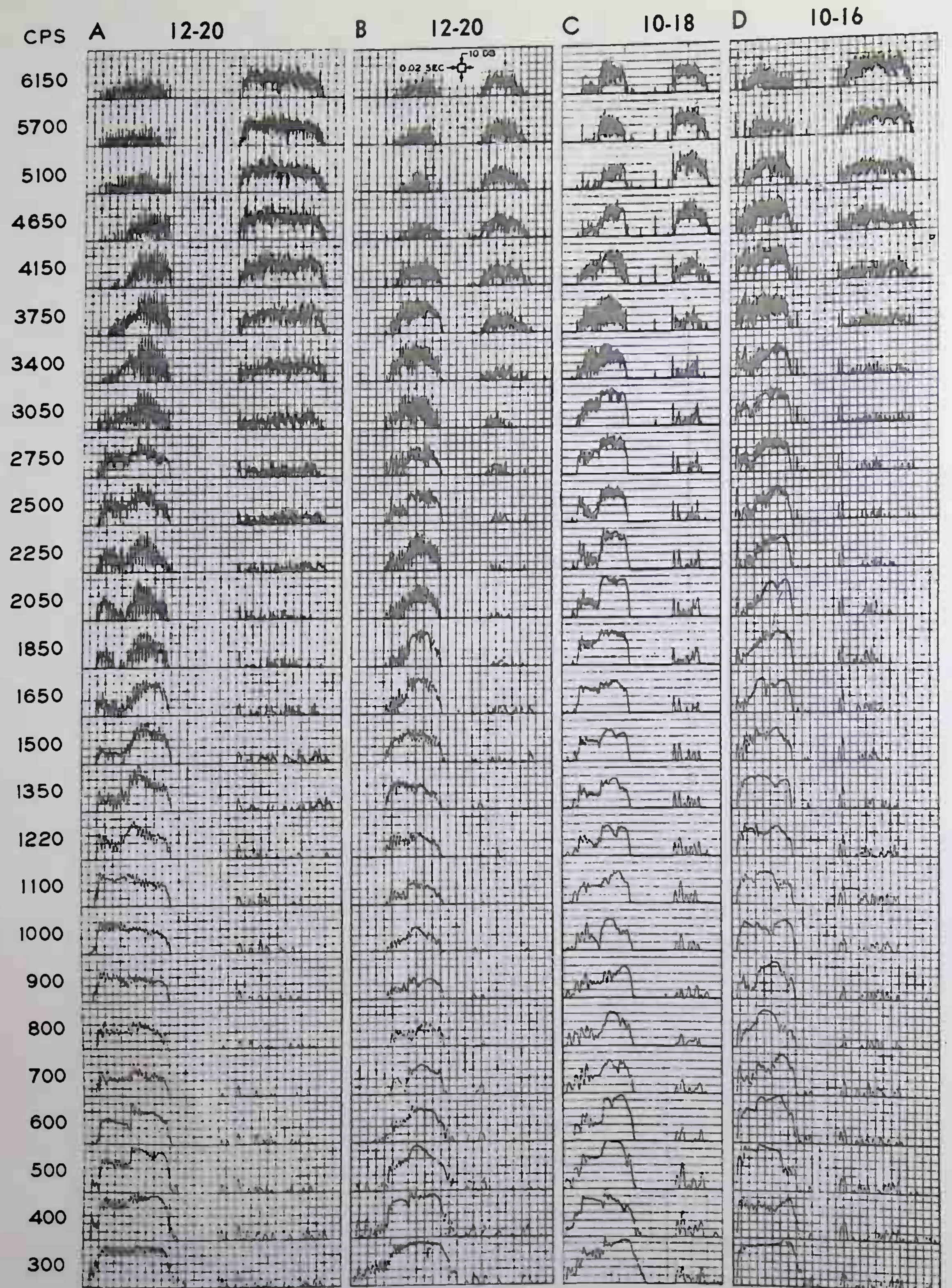
BLITZ



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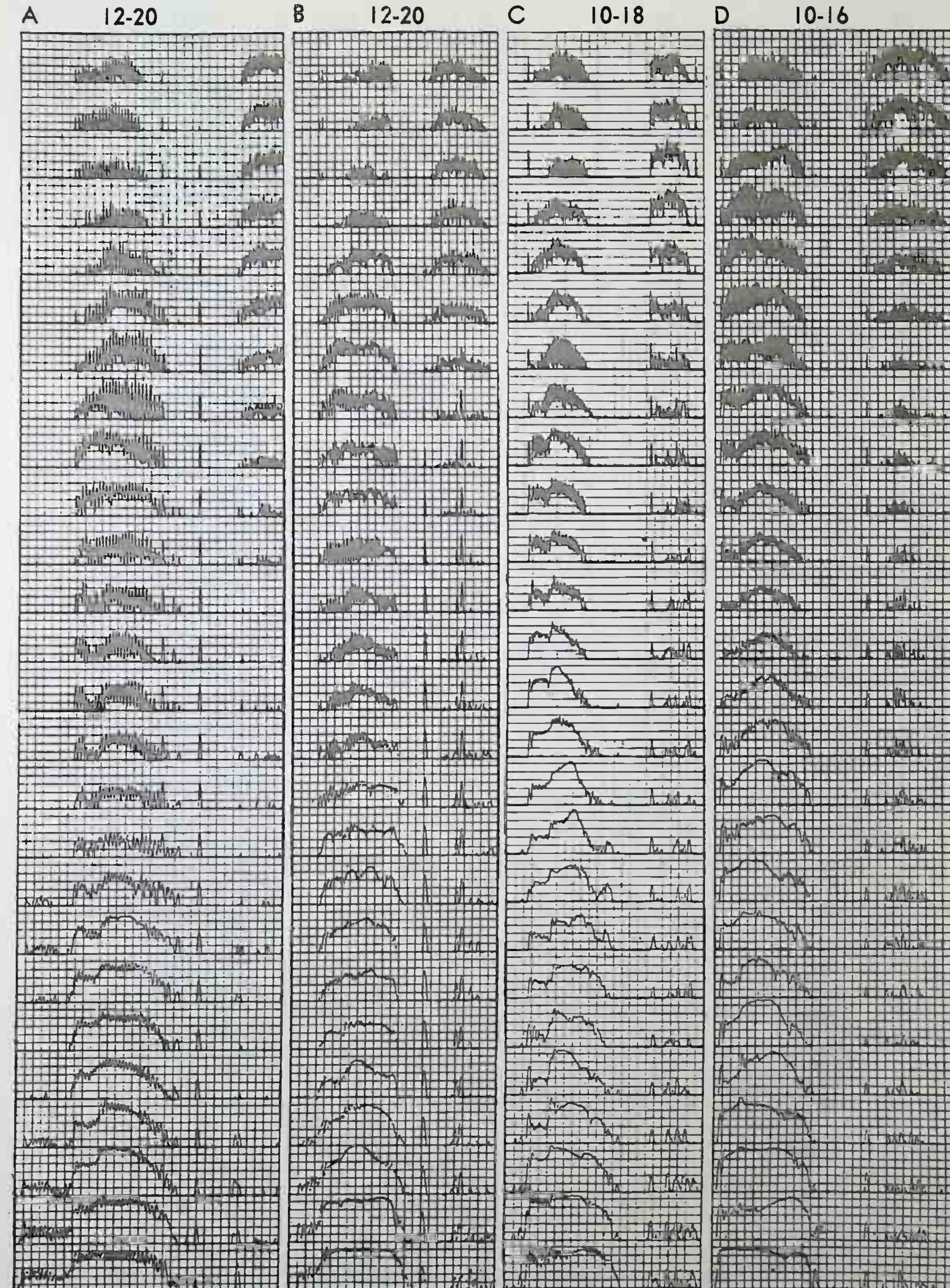

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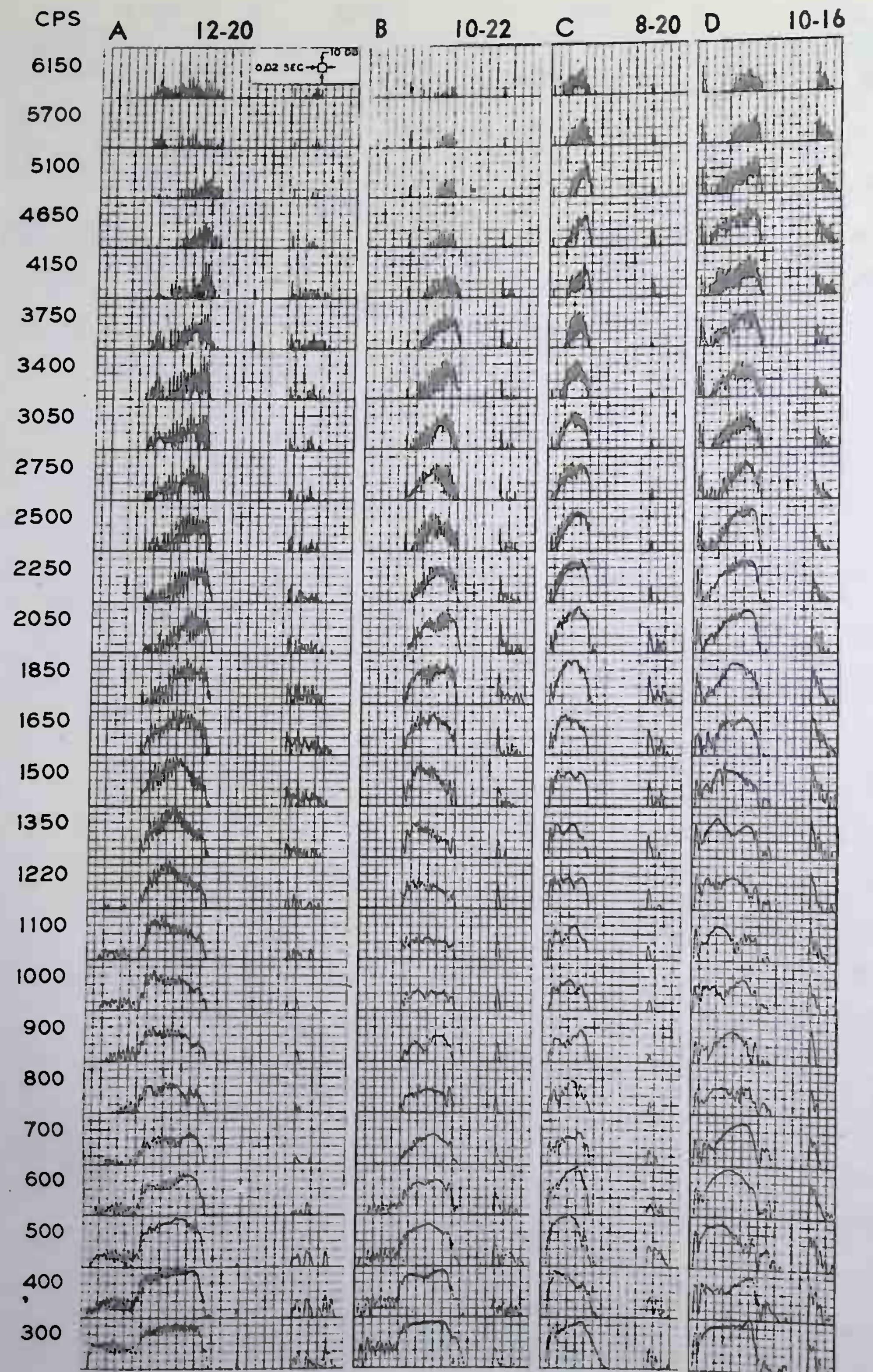
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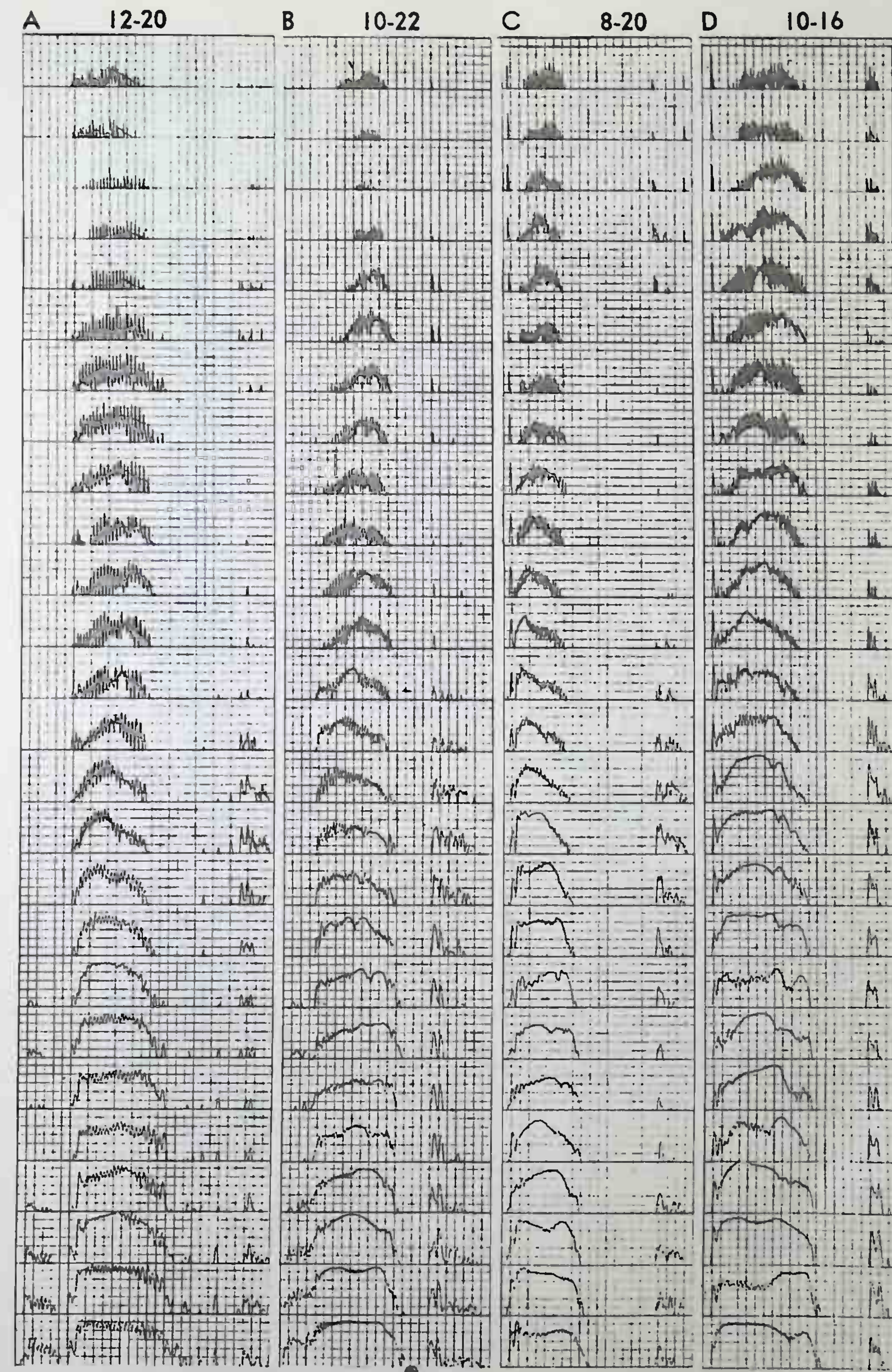
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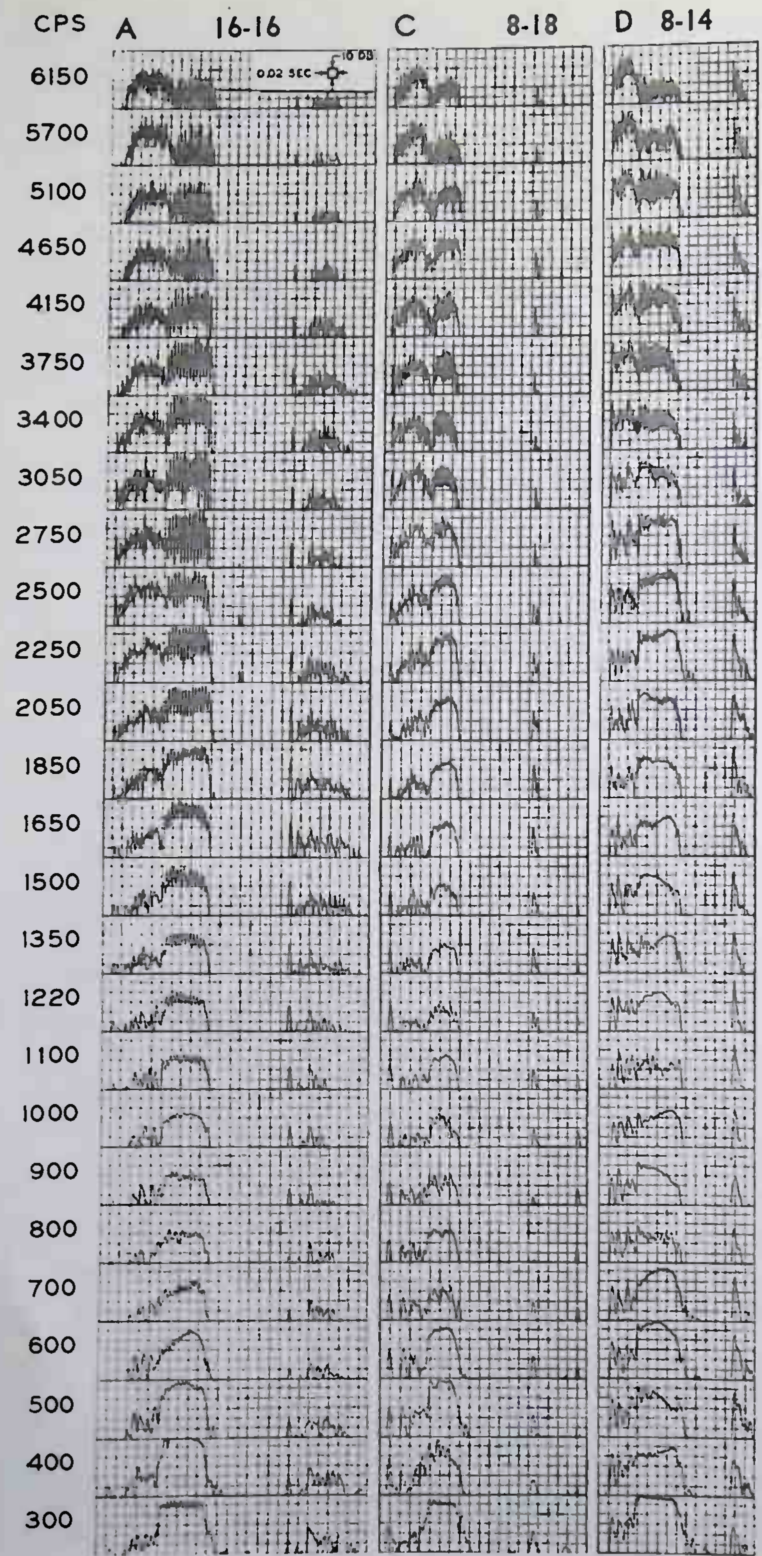
CHICK



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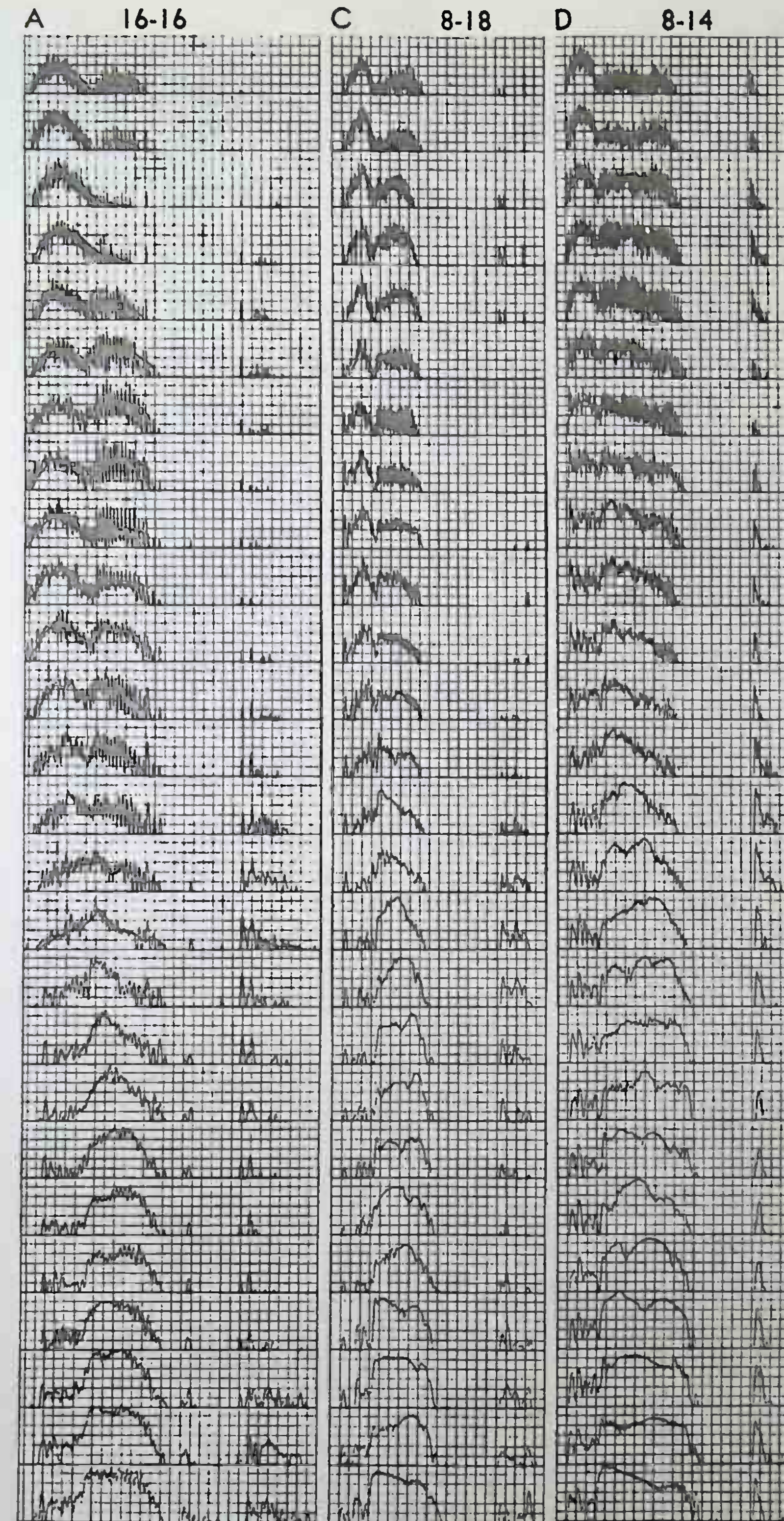

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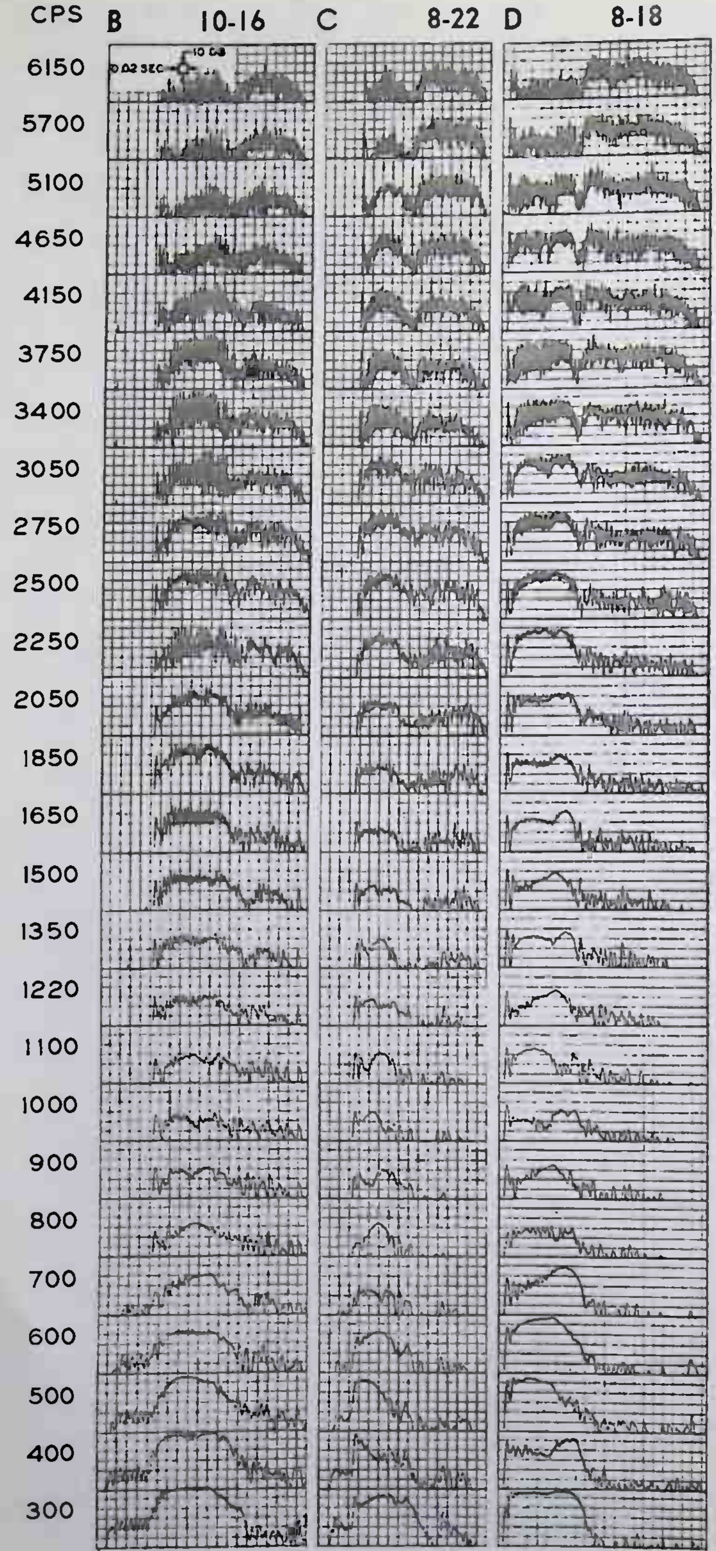
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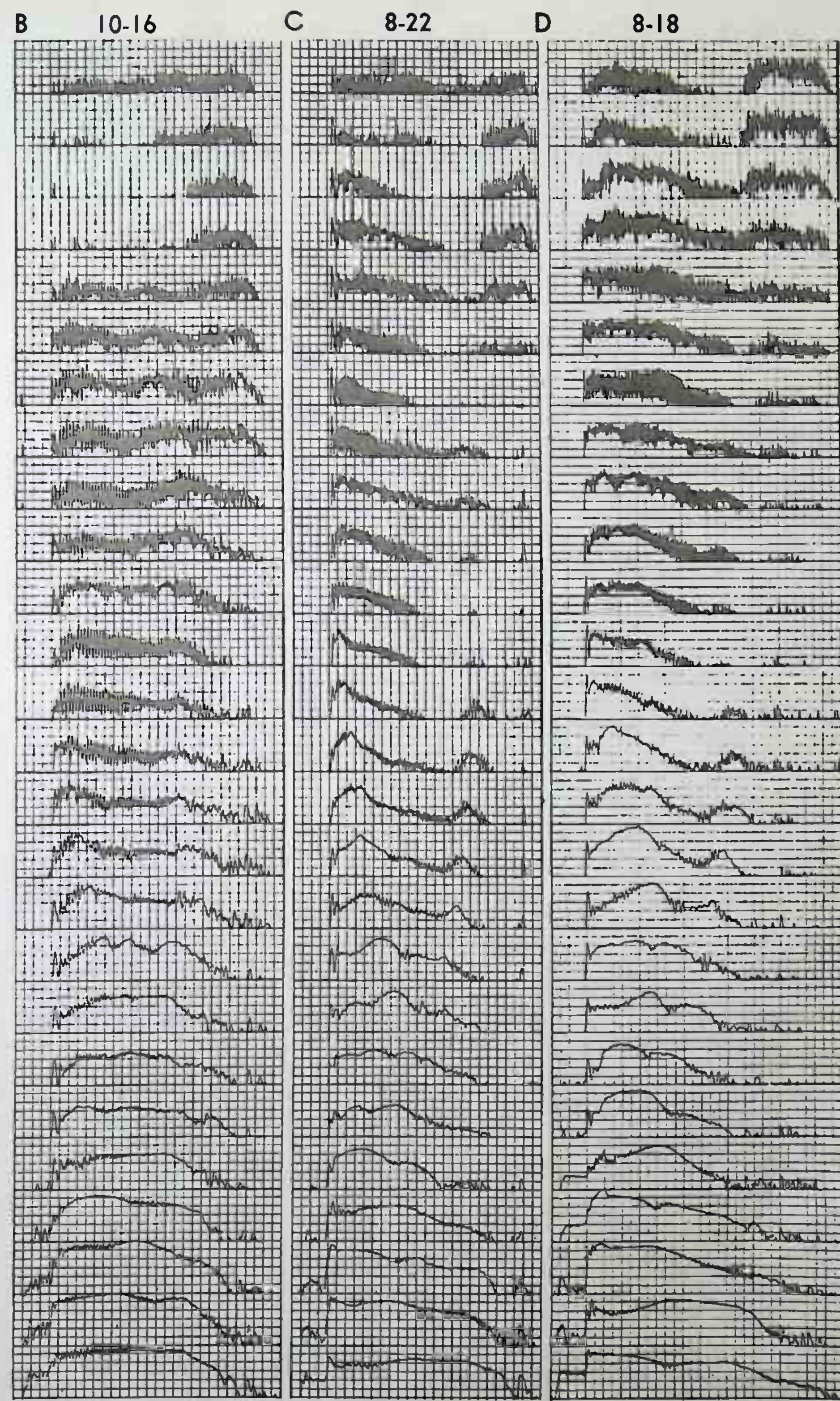
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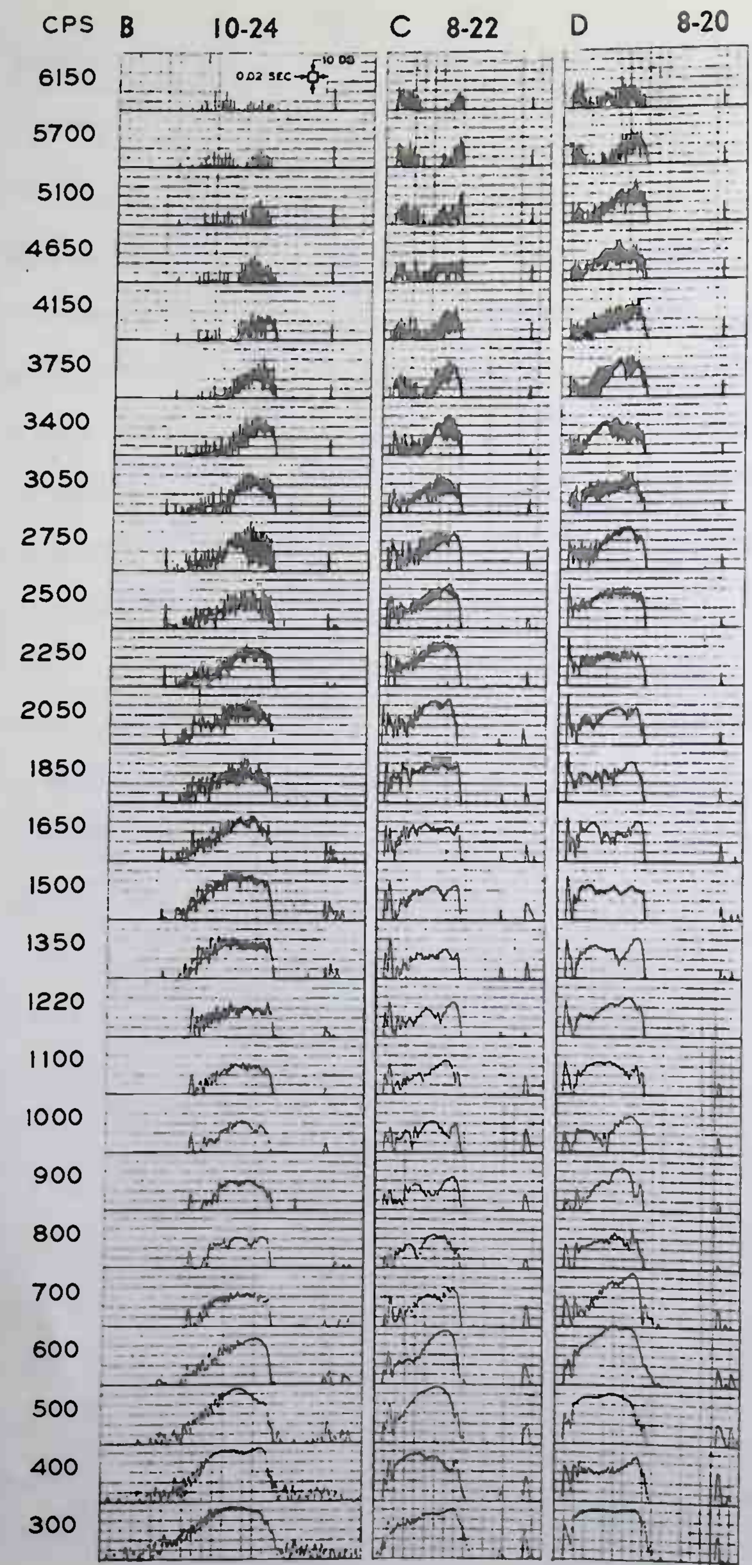
C 8-22
D 8-18

DRIP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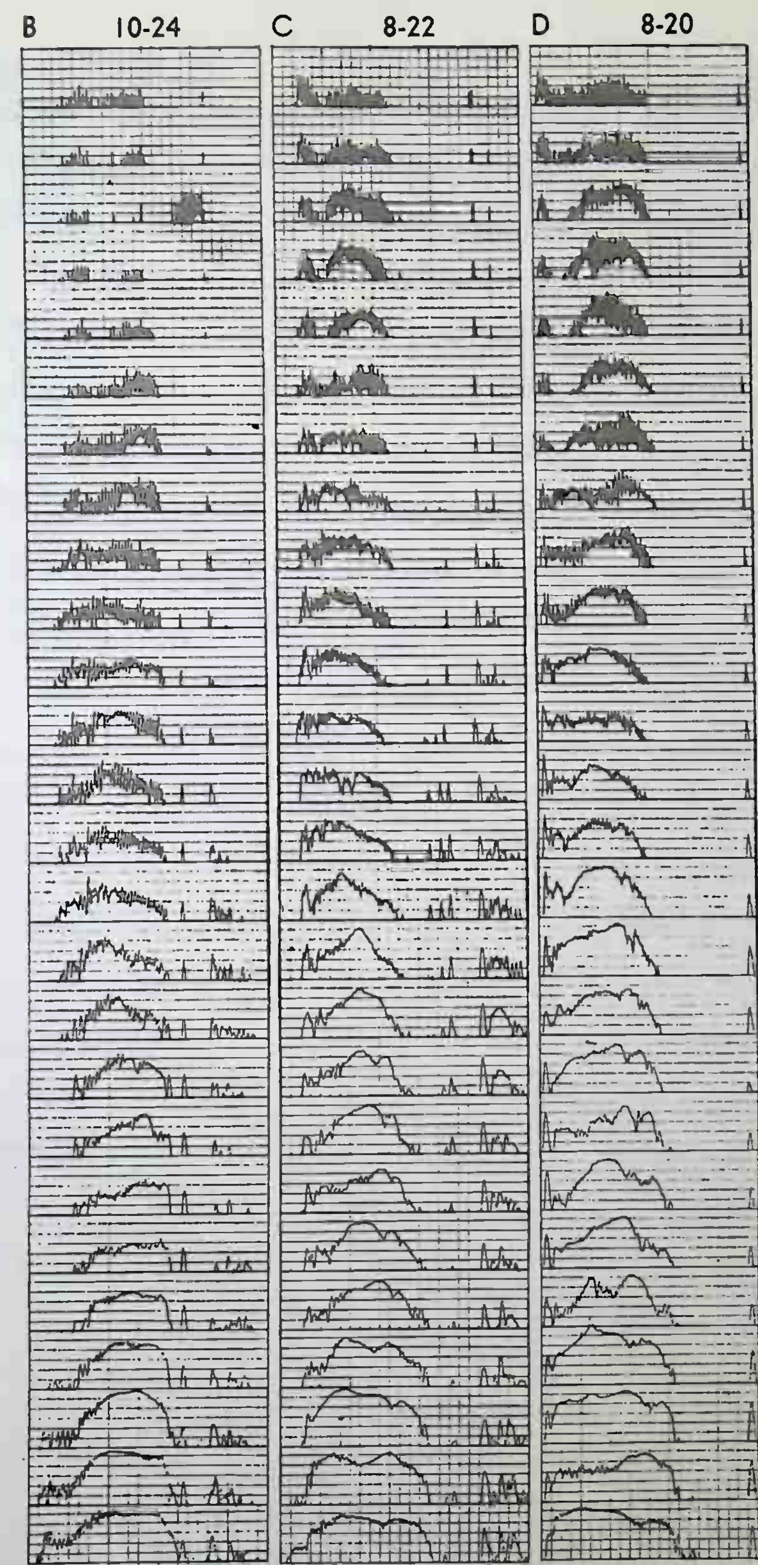
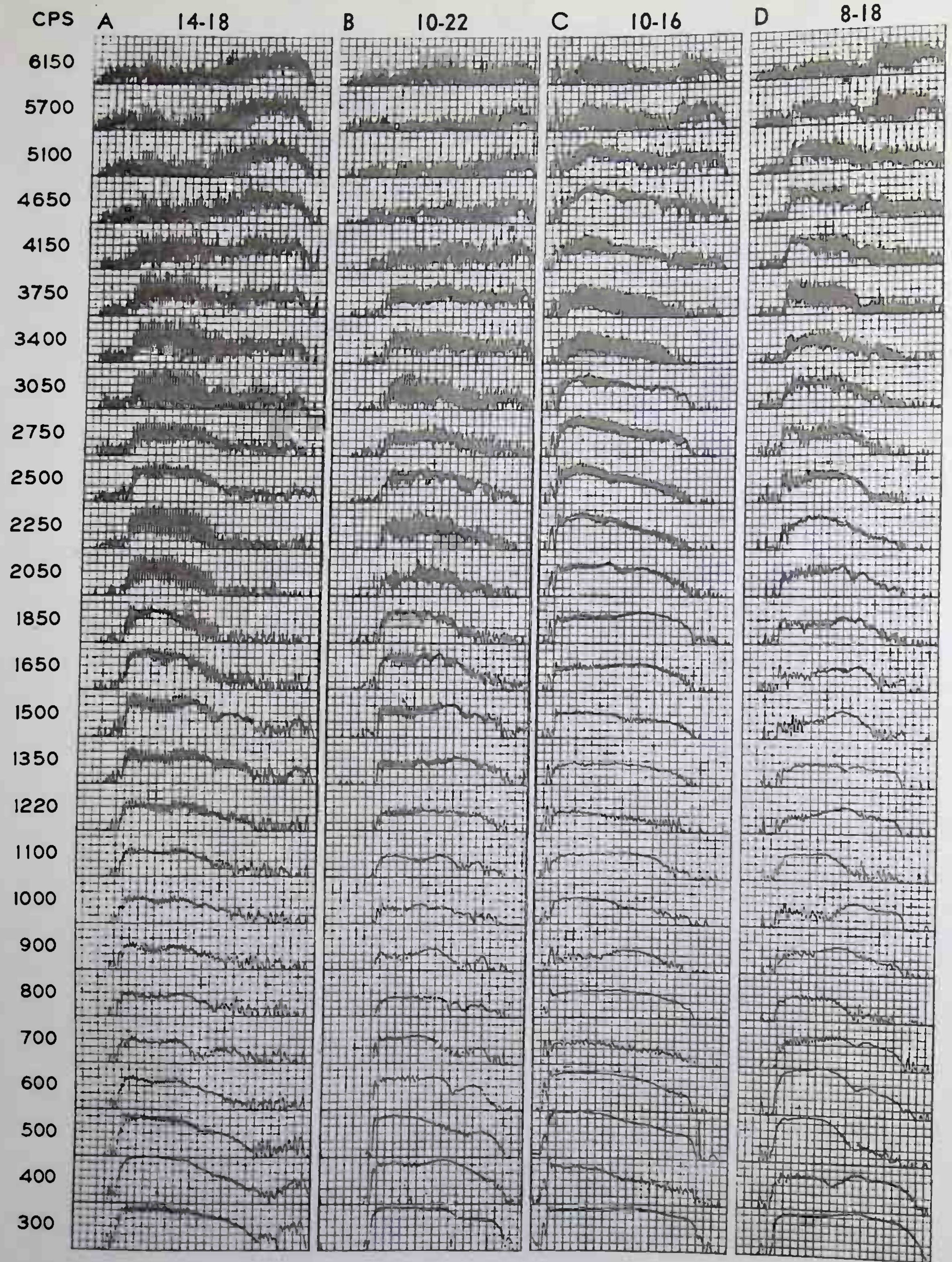

B 10-24
C 8-22
D 8-20

FIZZ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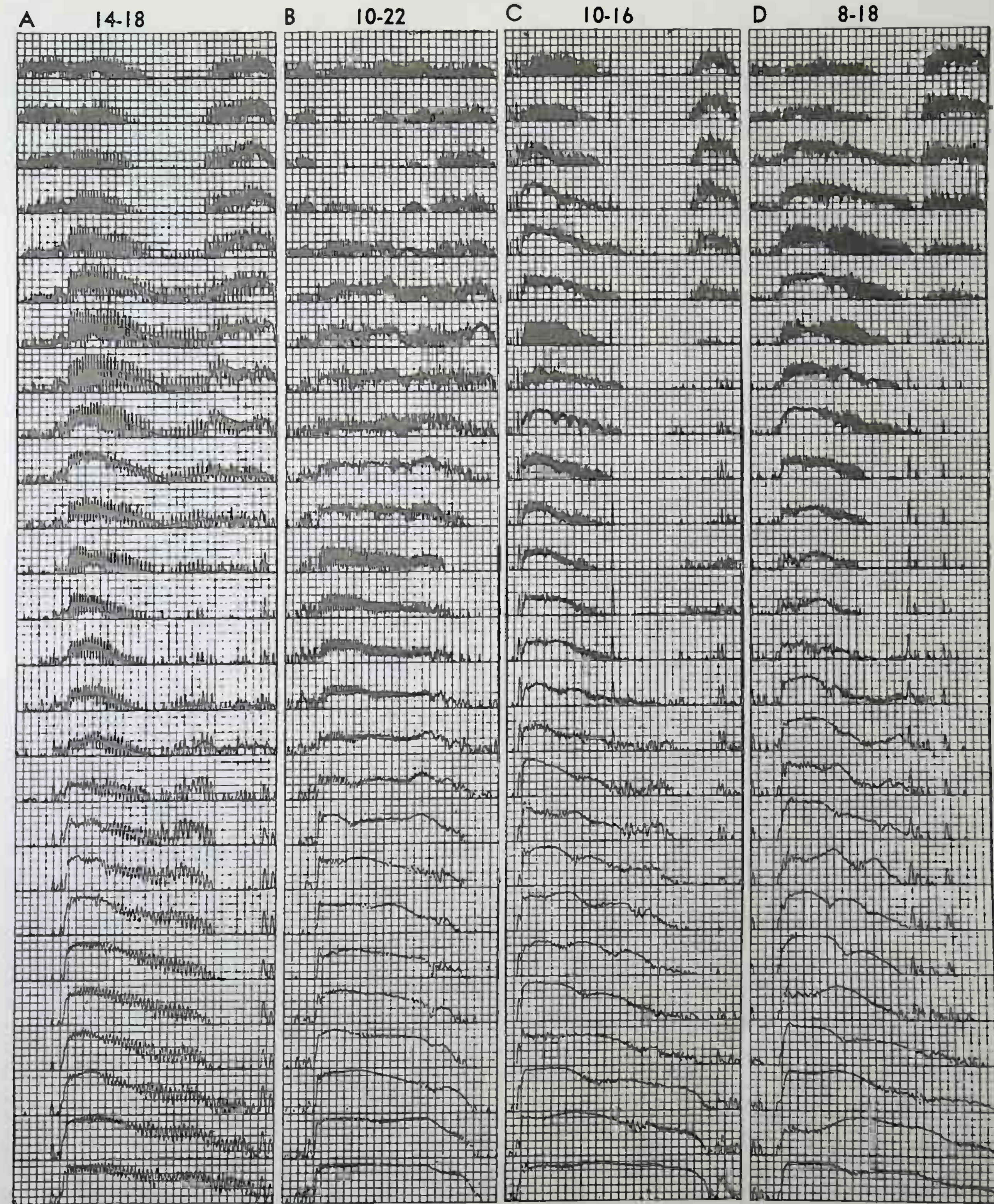
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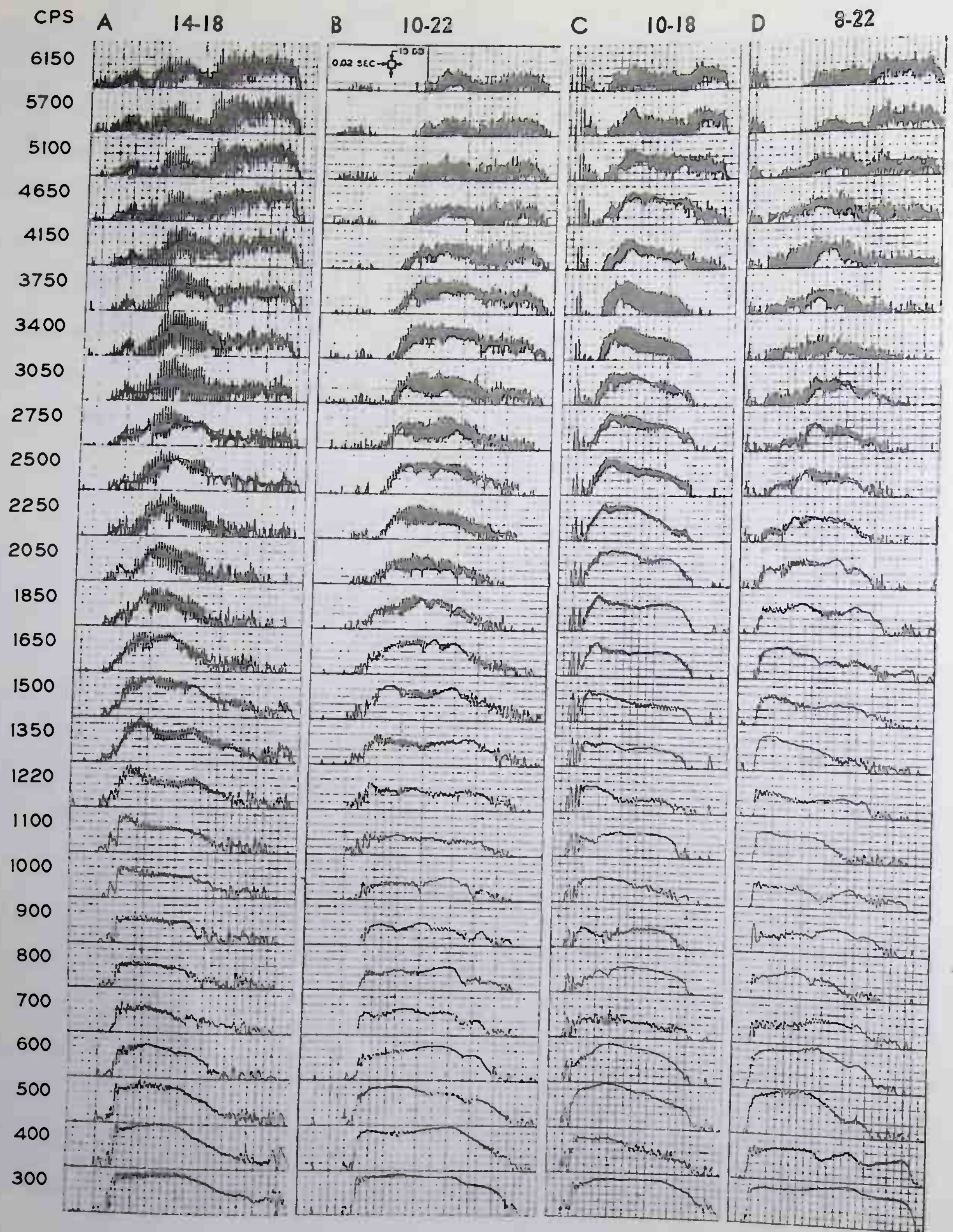
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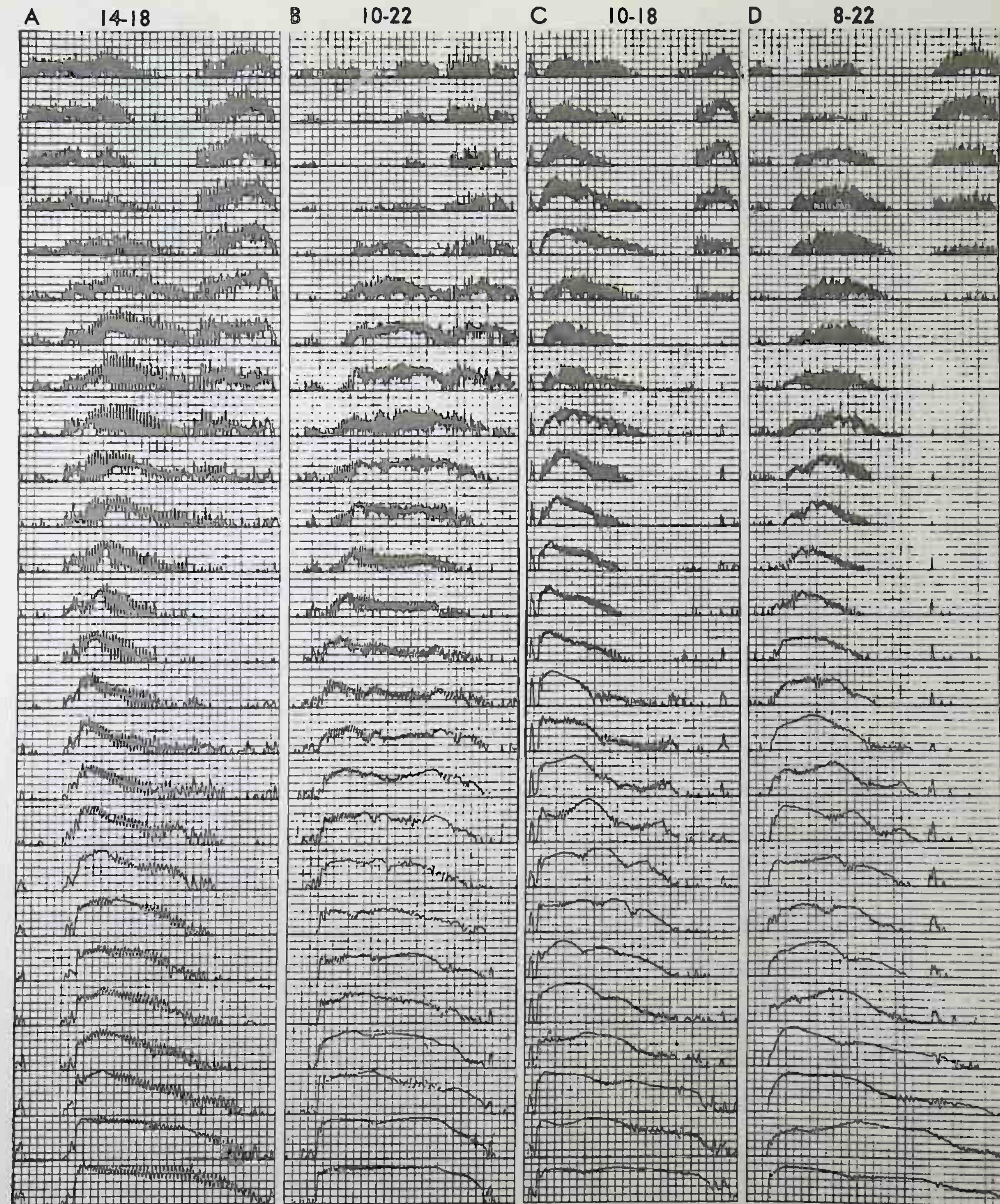
FROZE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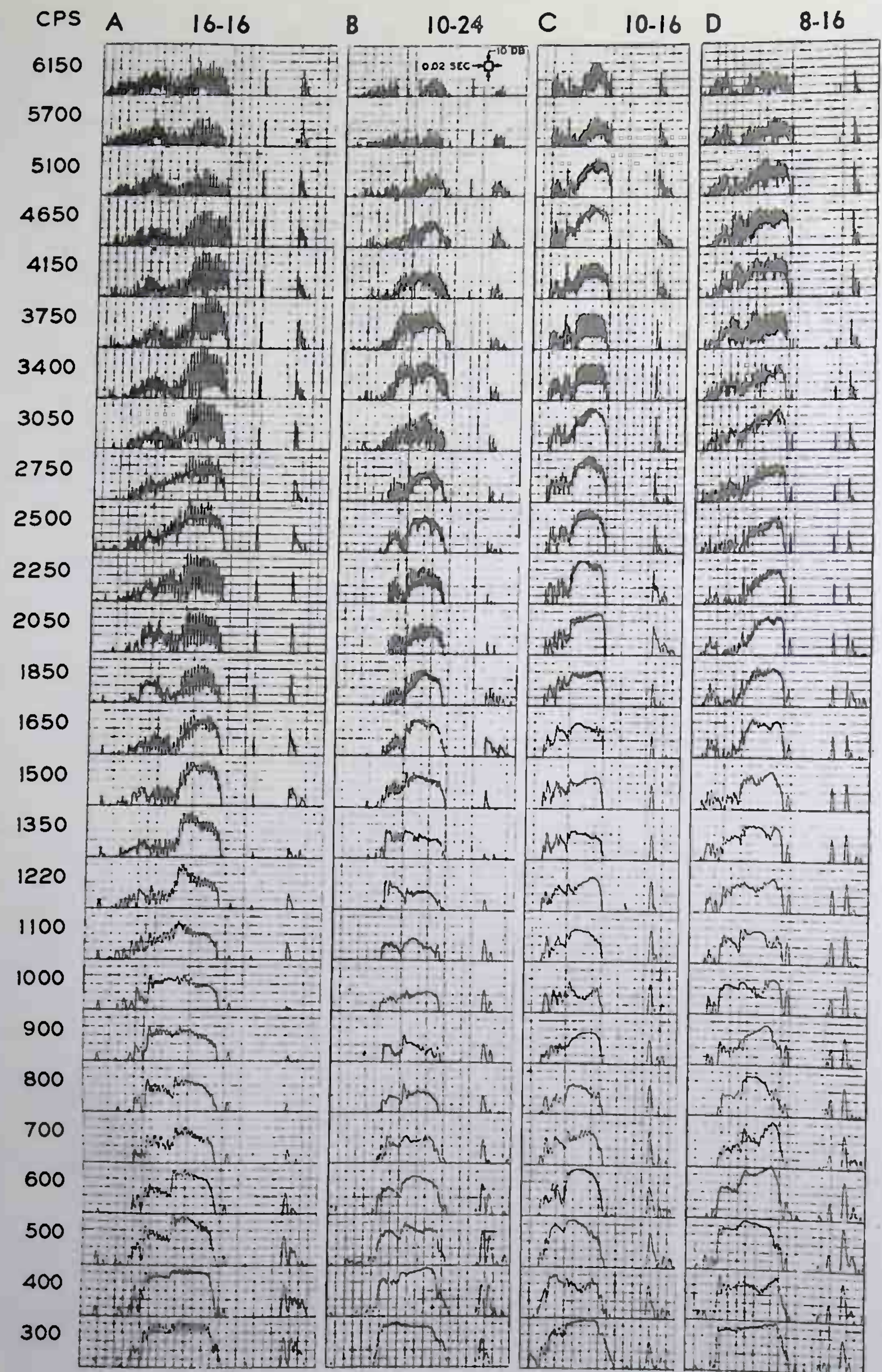
D 8-22

FLIT


CPS
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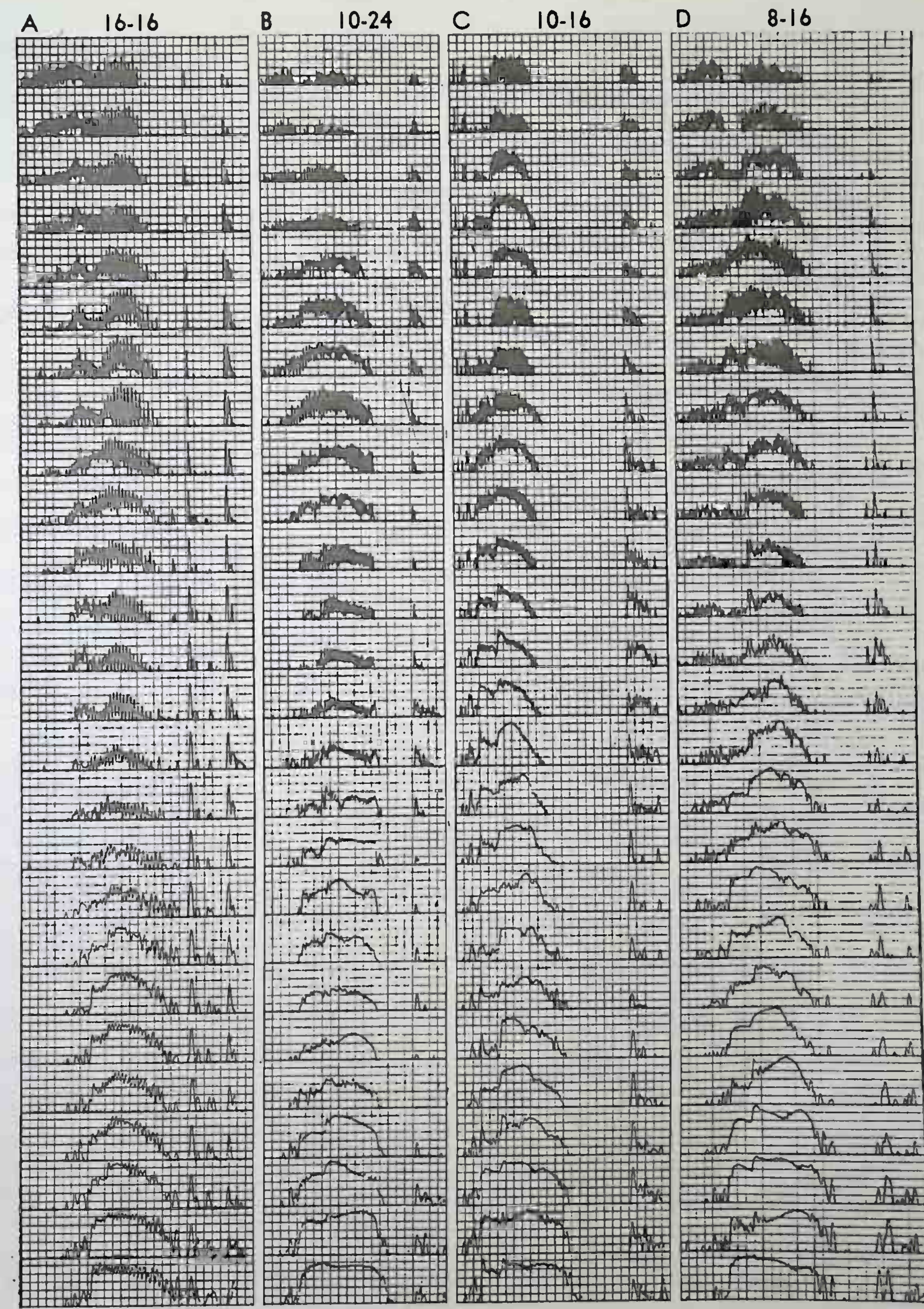
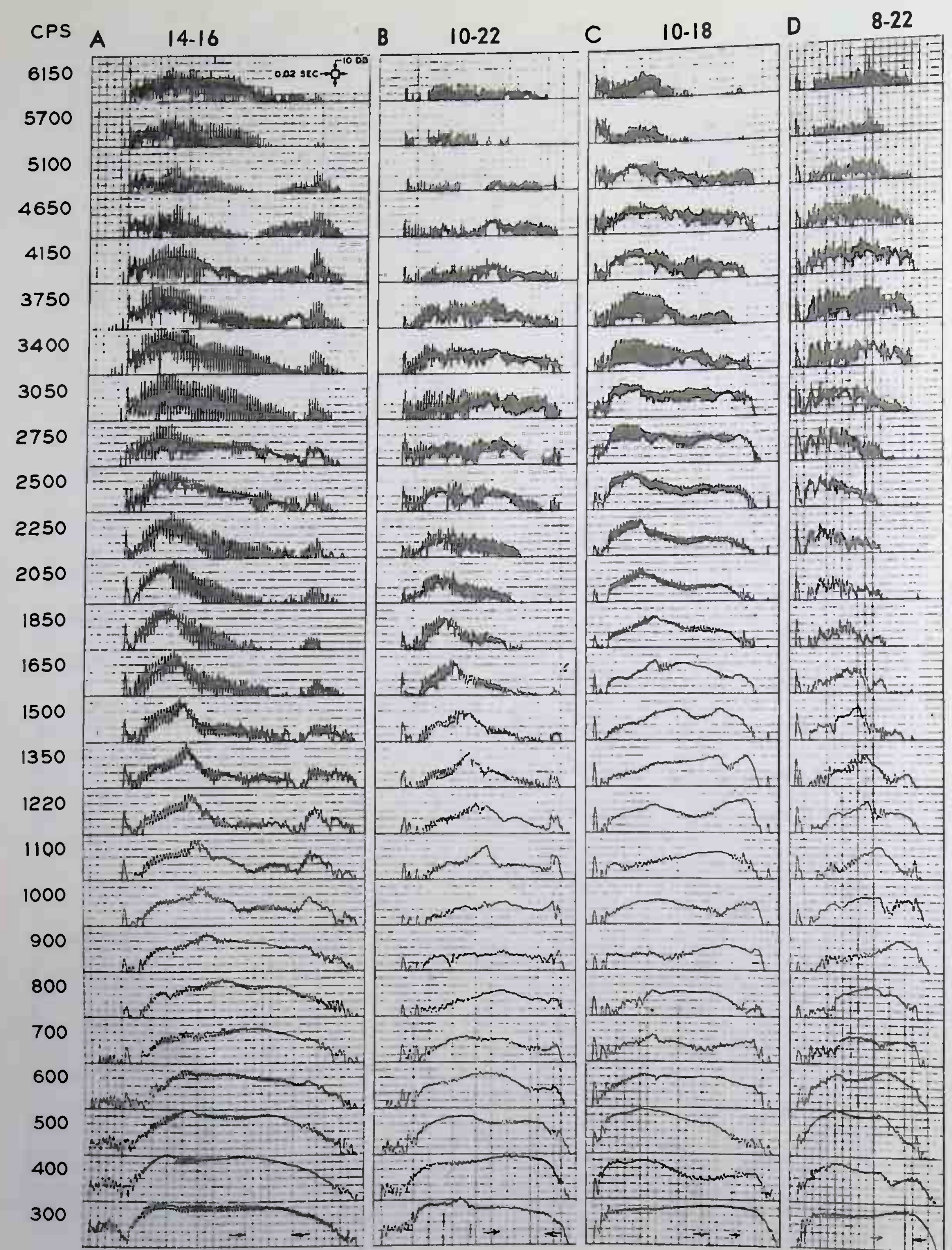

A 16-16
B 10-24
C 10-16
D 8-16

GILL


CPS
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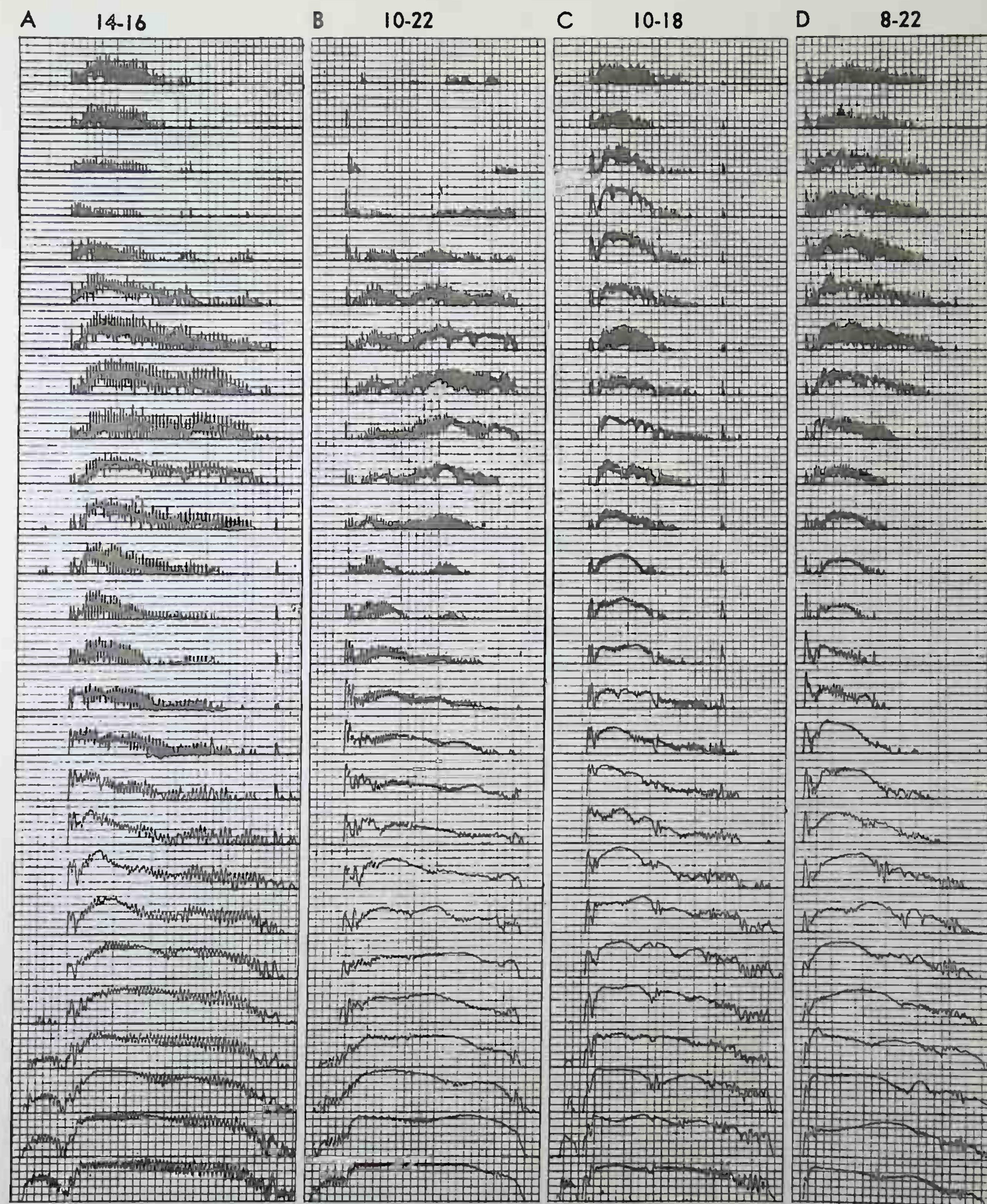
GOAL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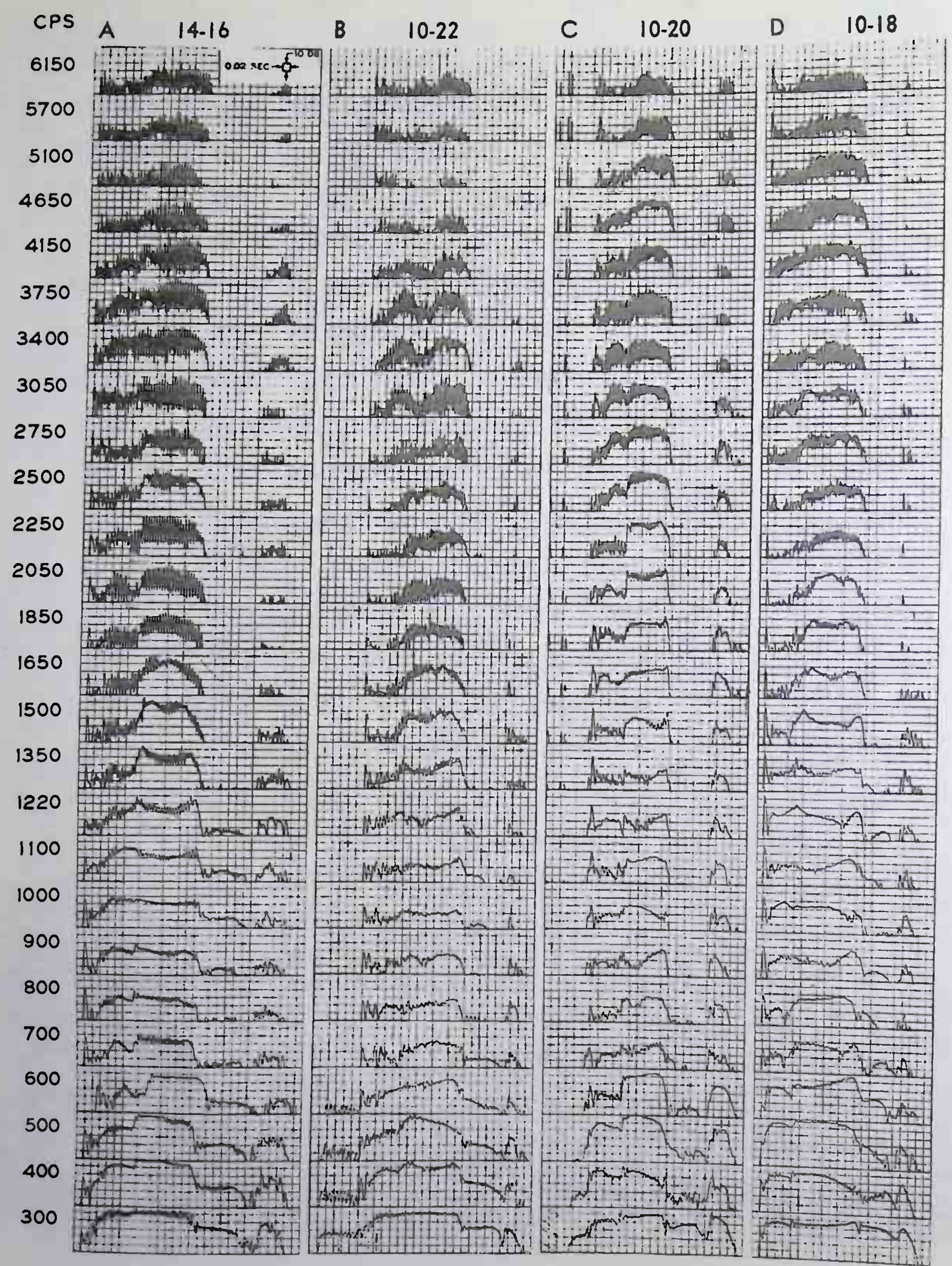
D 8-22

GLIB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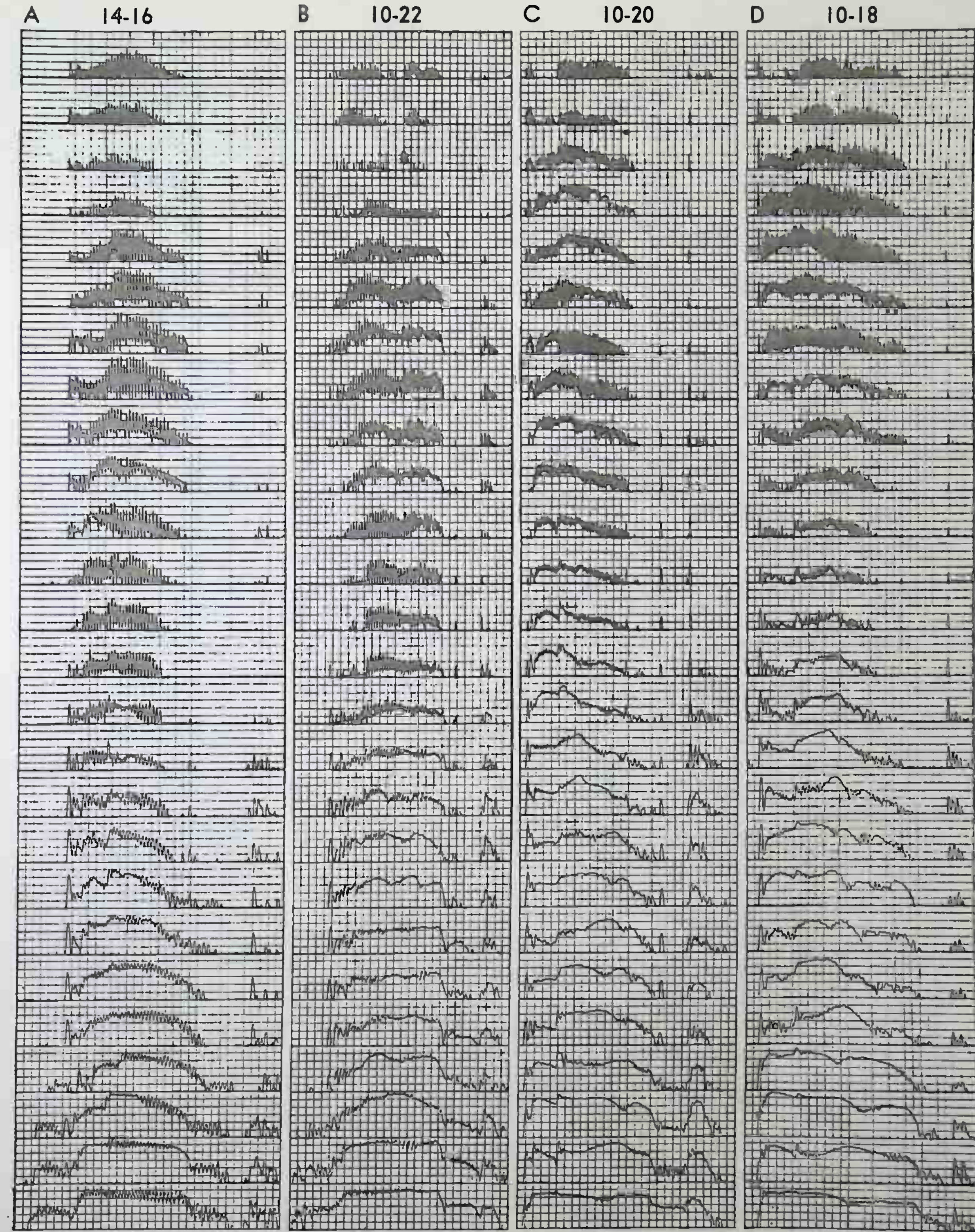
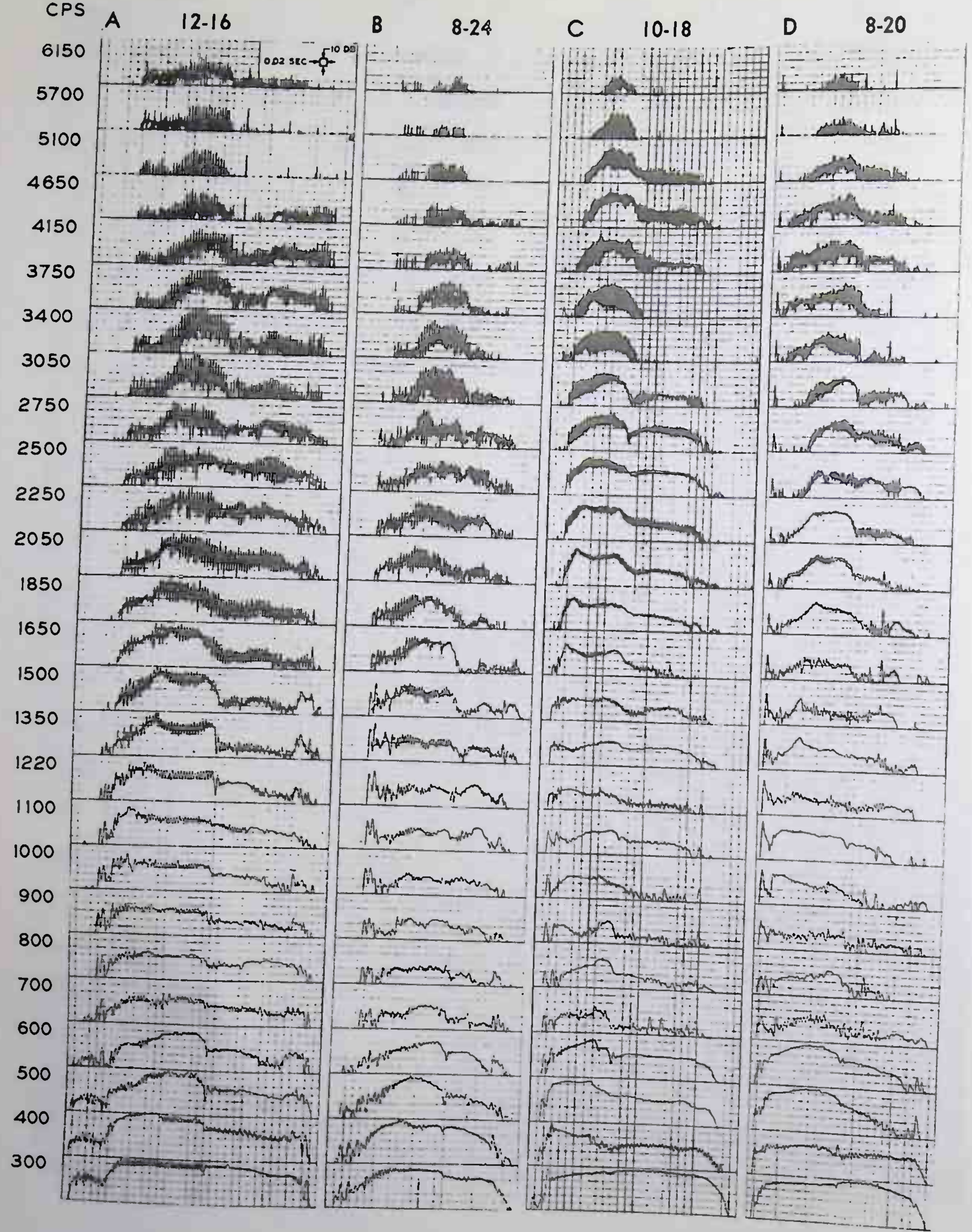
GLOBE


A 14-16
B 10-22
C 10-20
D 10-18

GRIN



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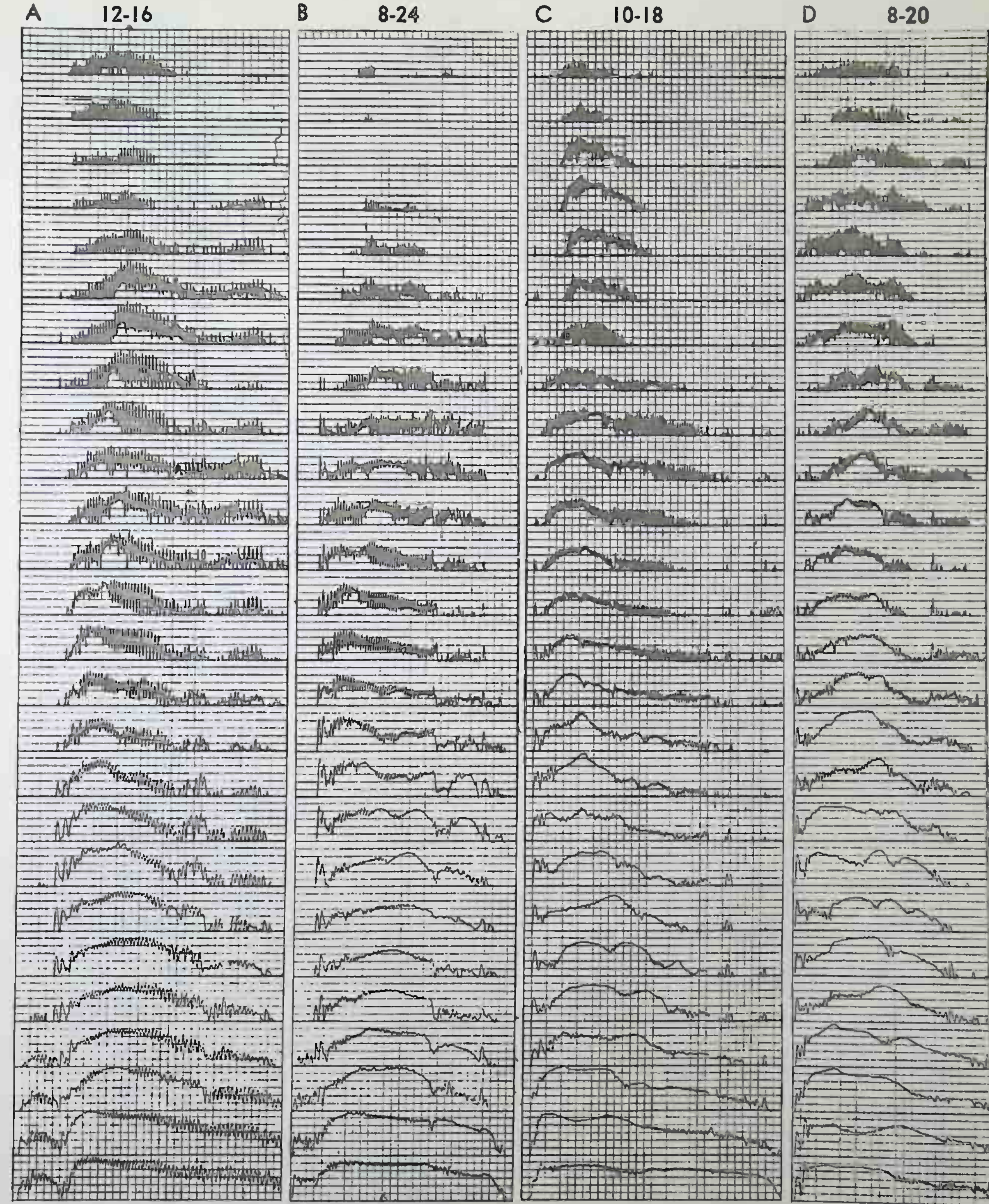

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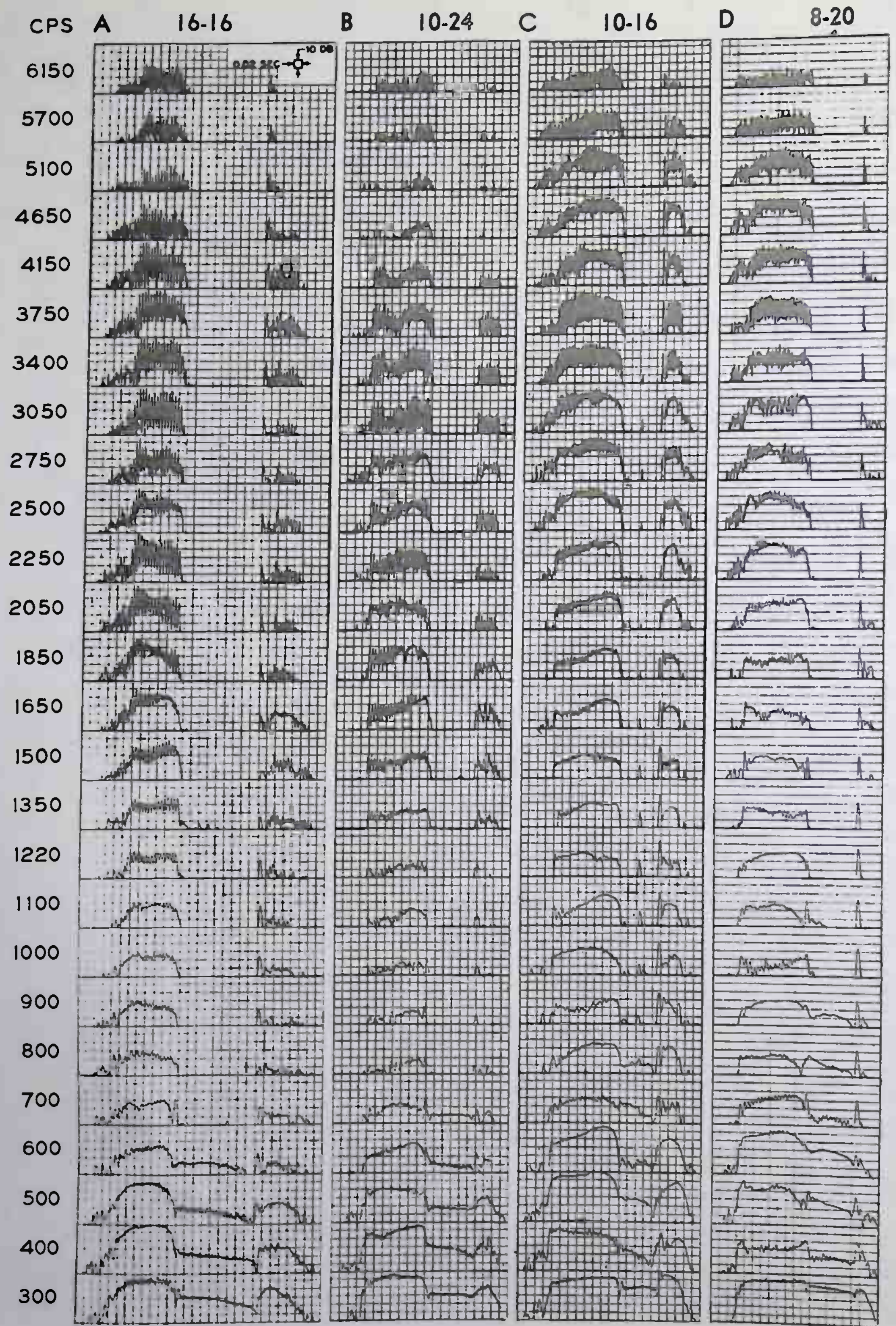

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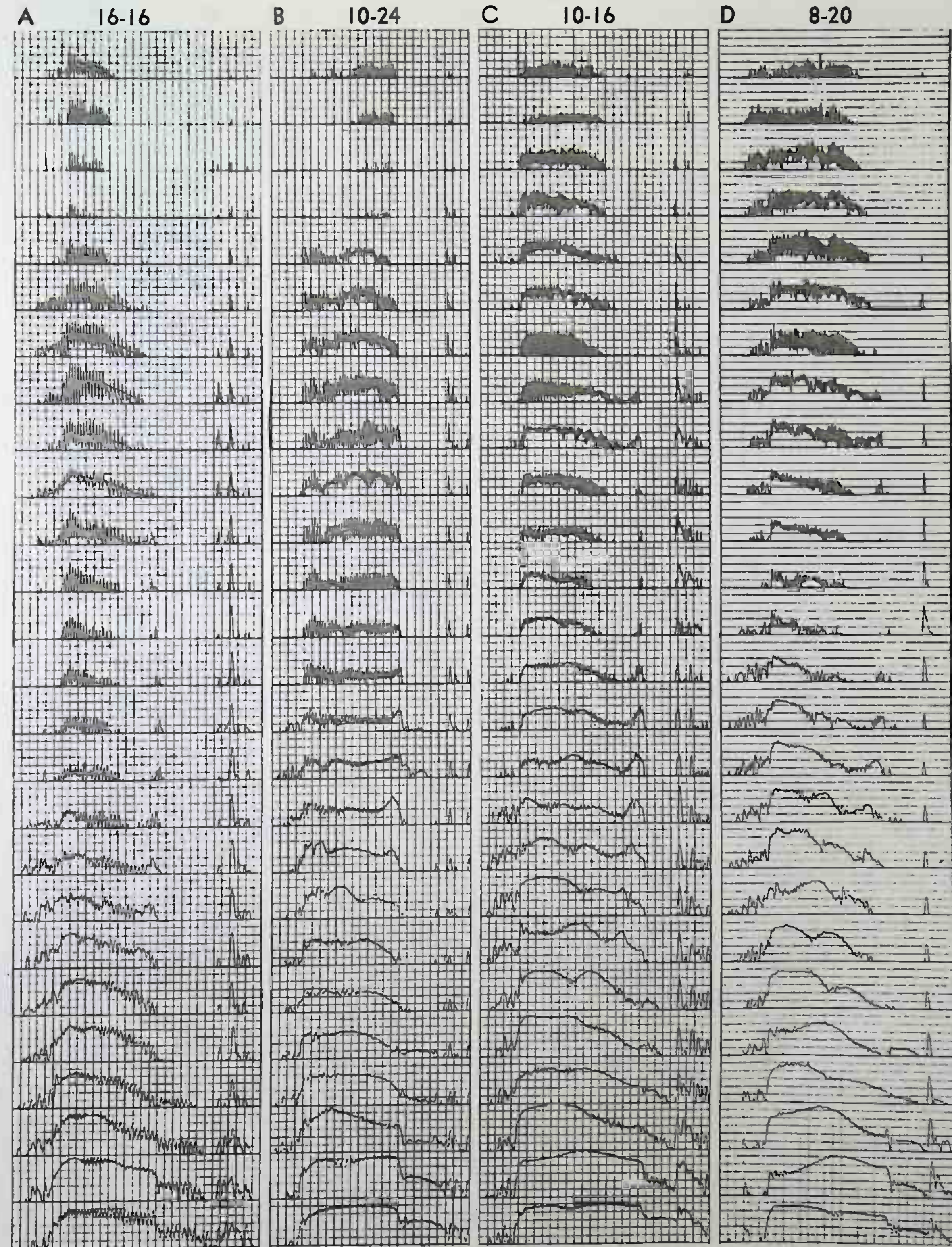

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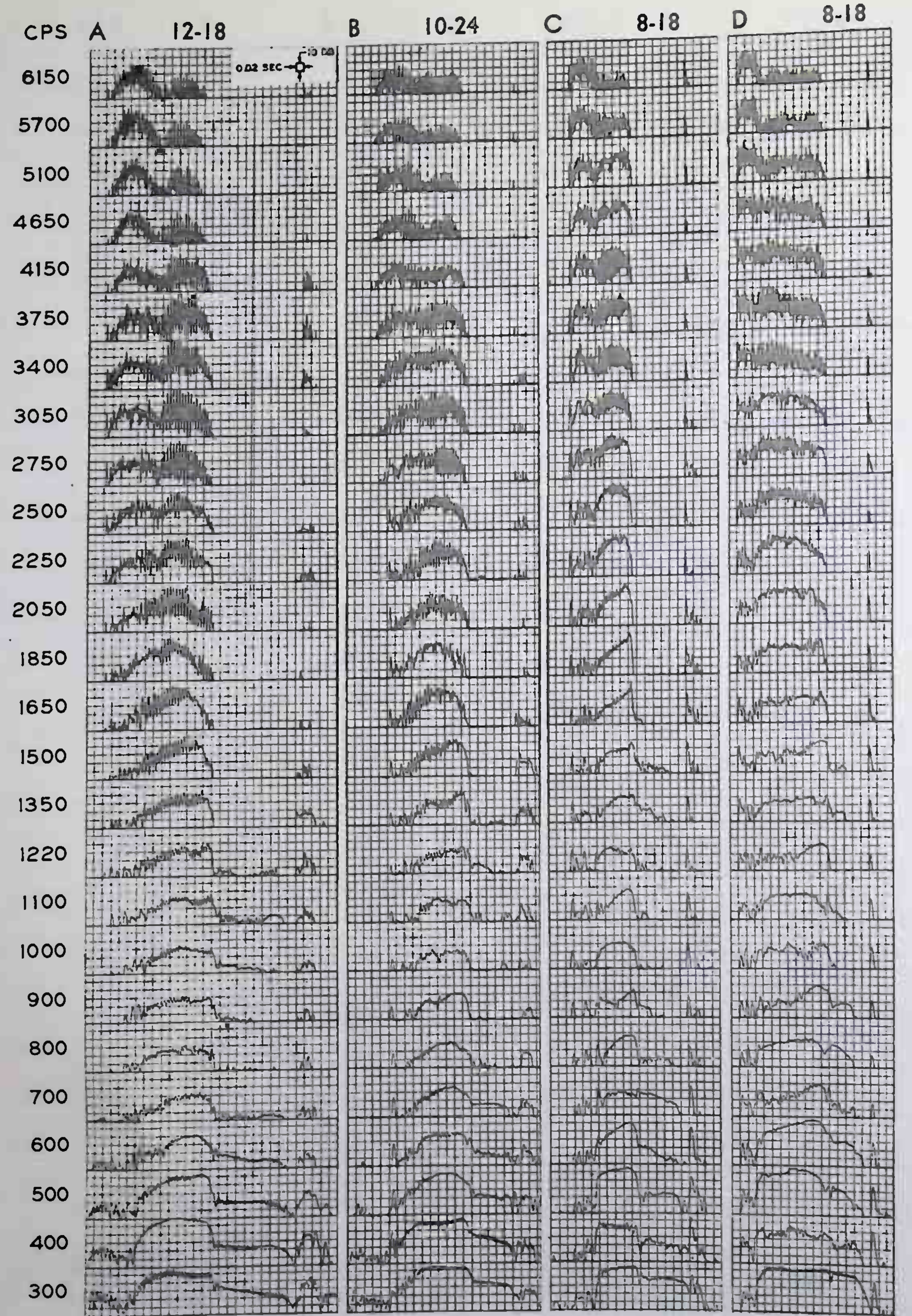
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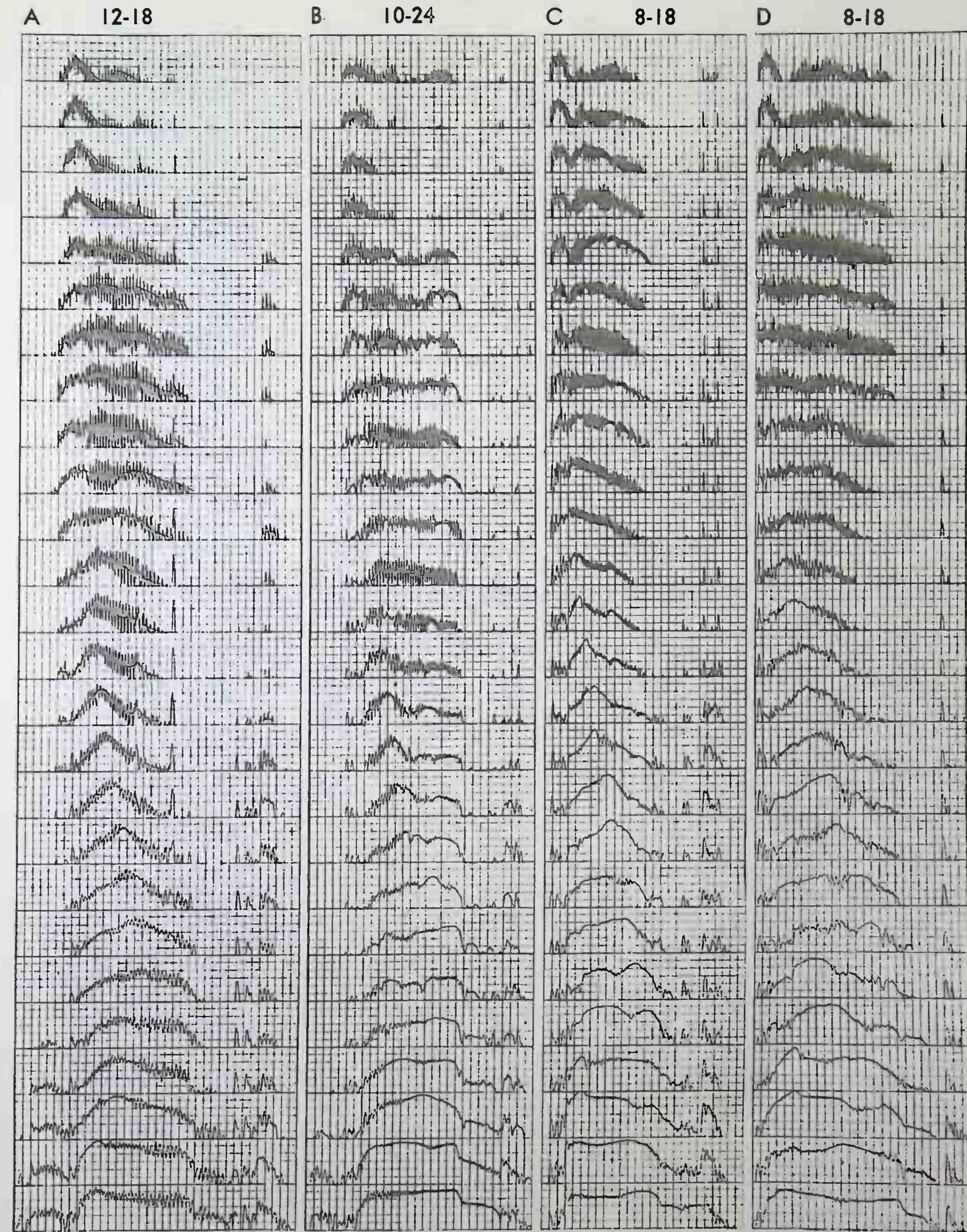
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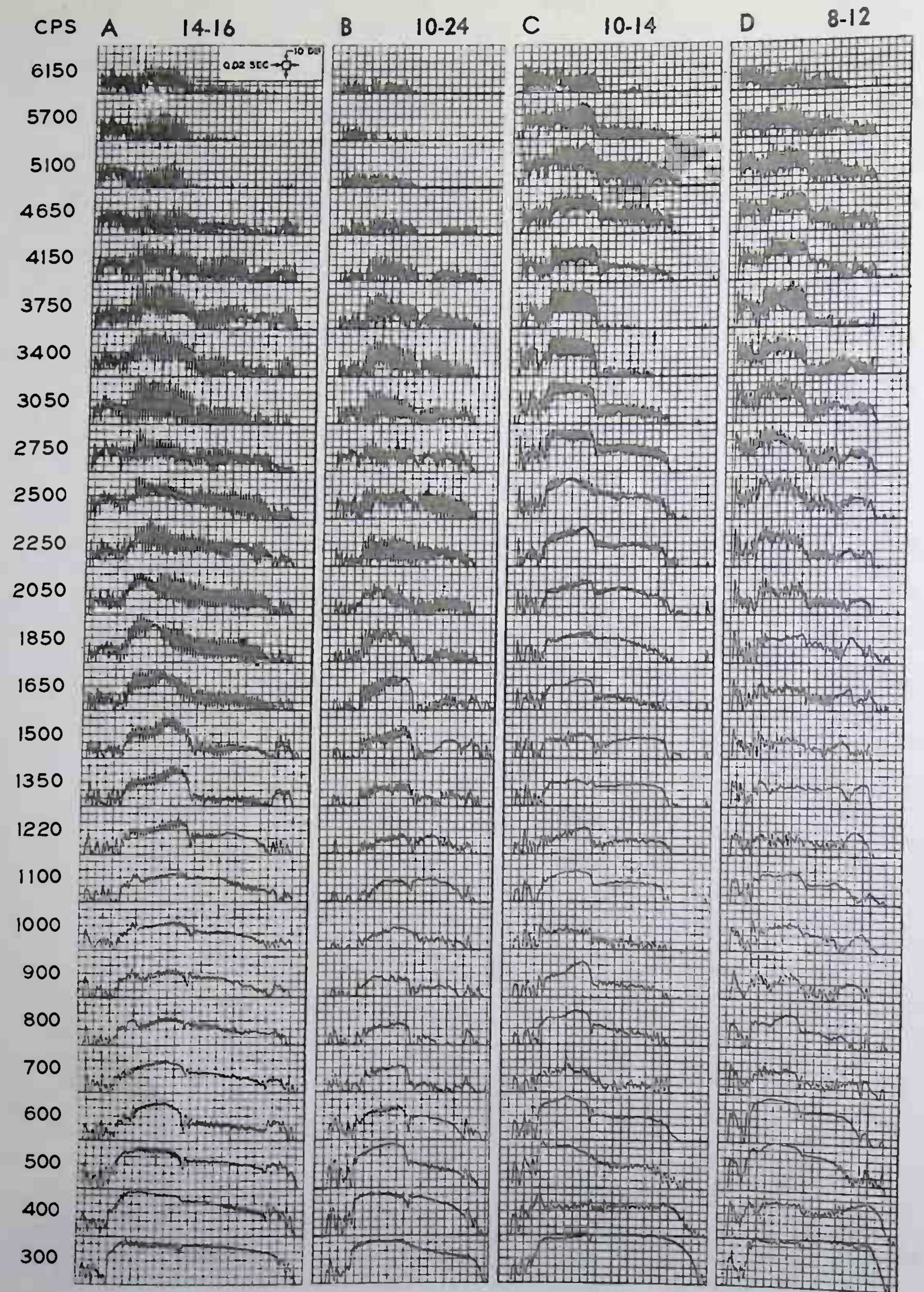
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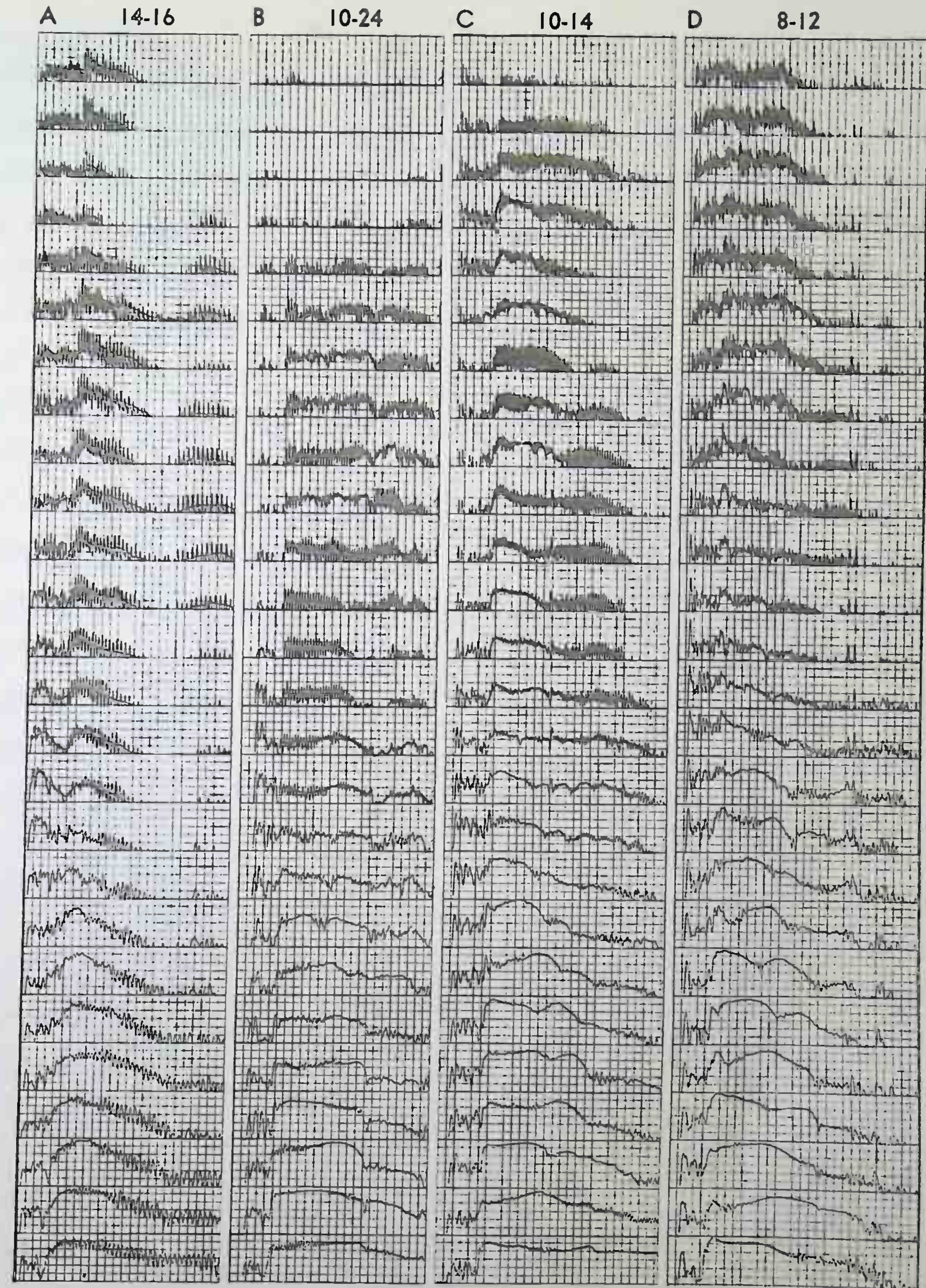
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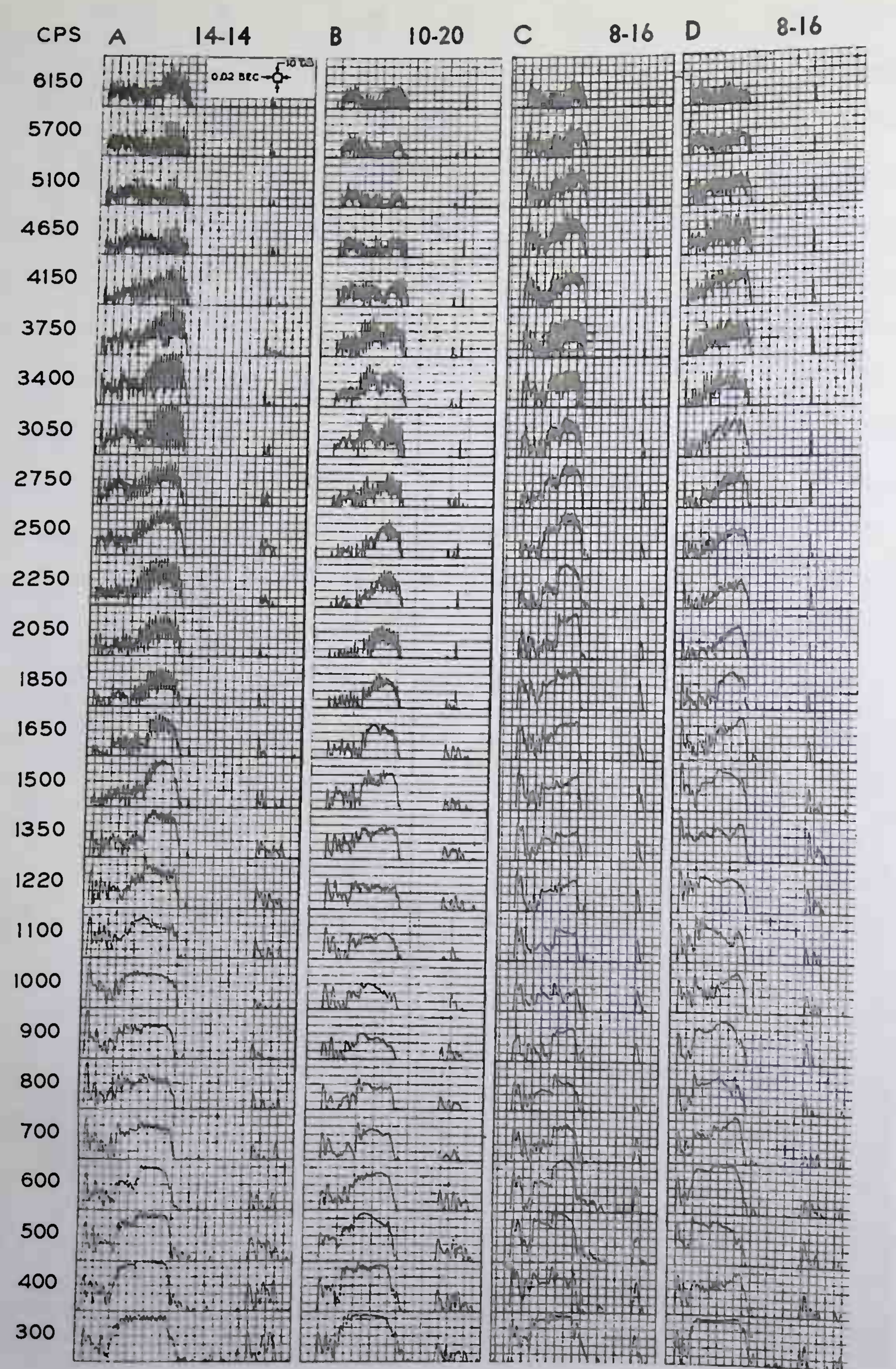
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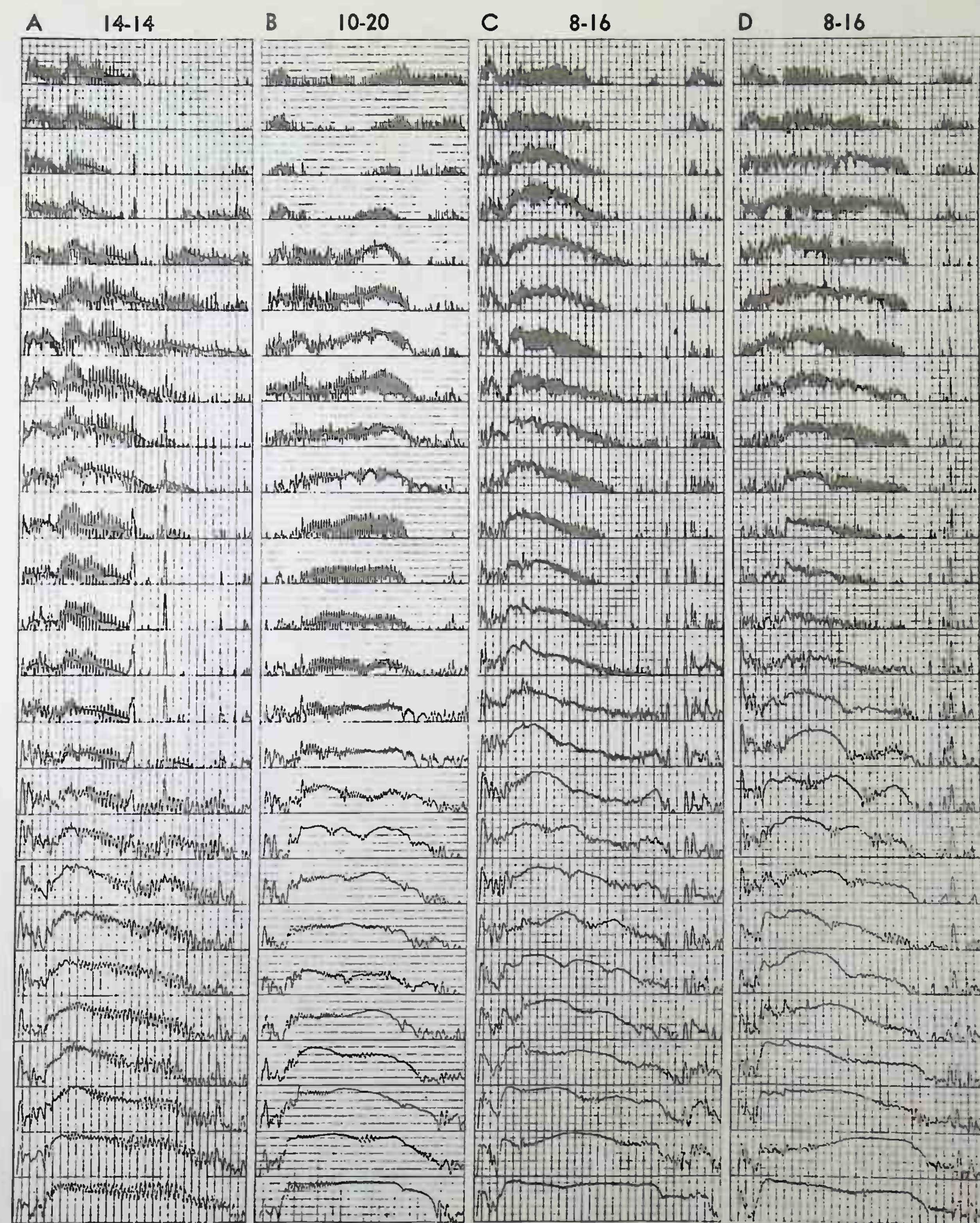
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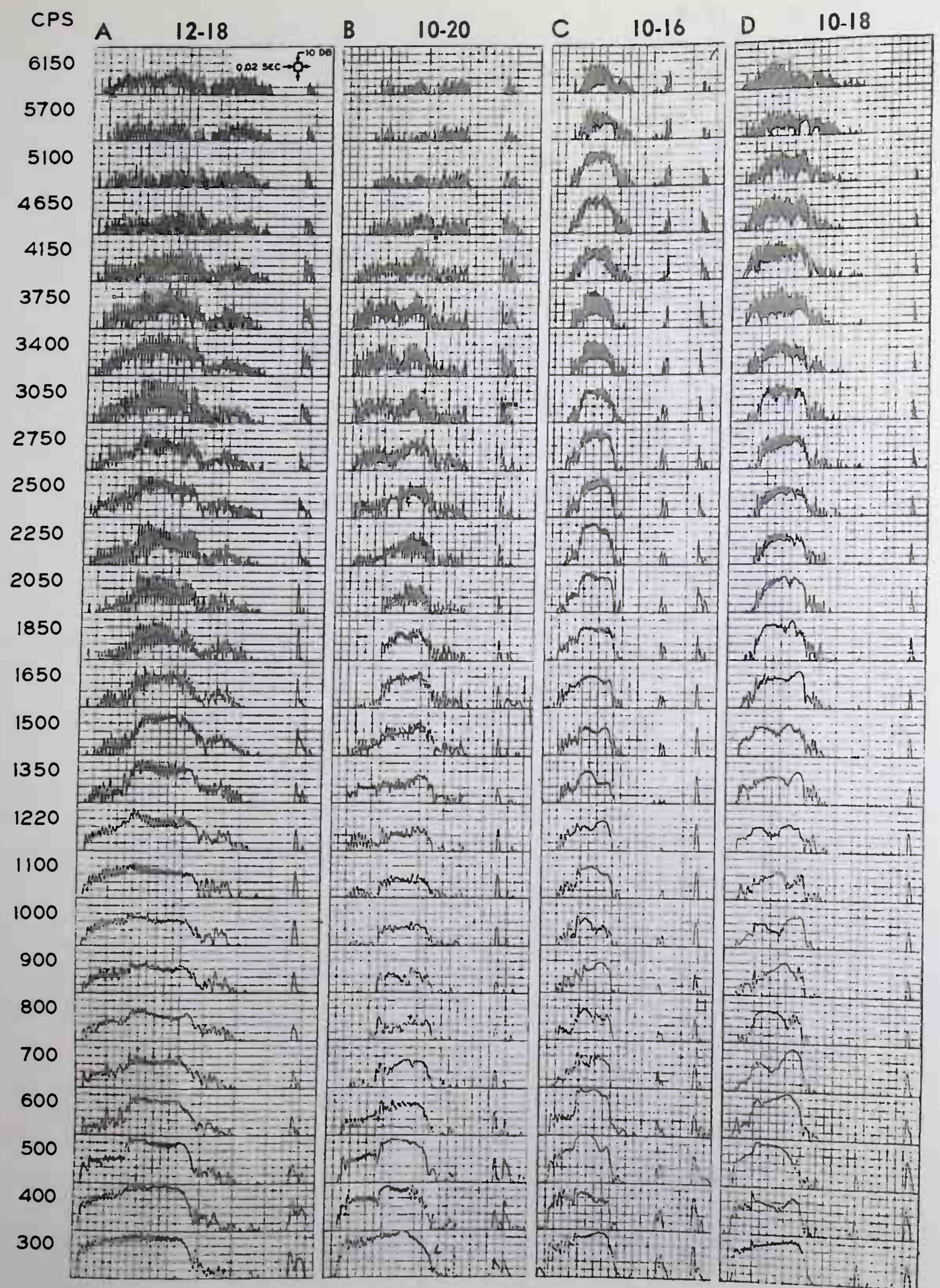
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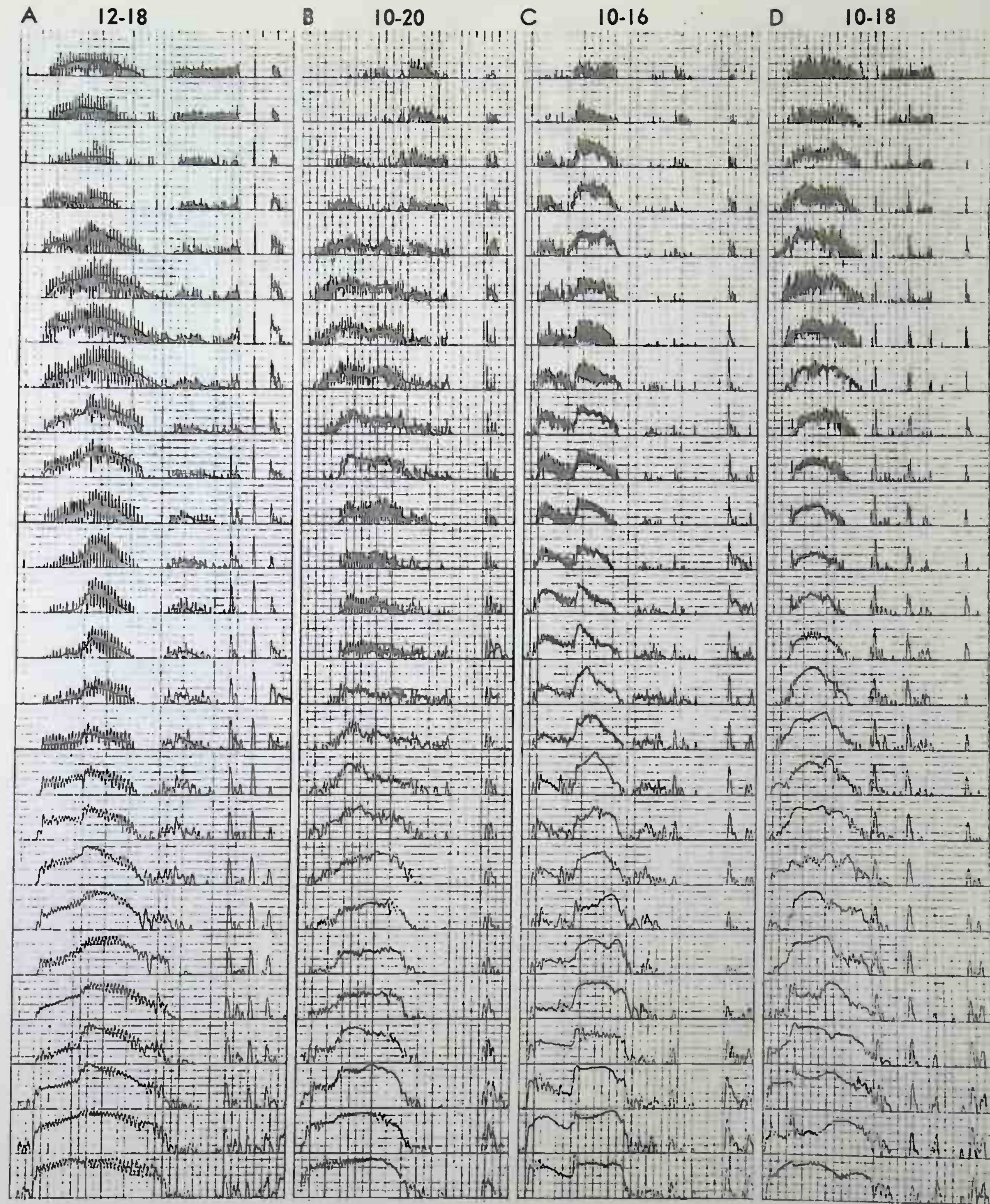
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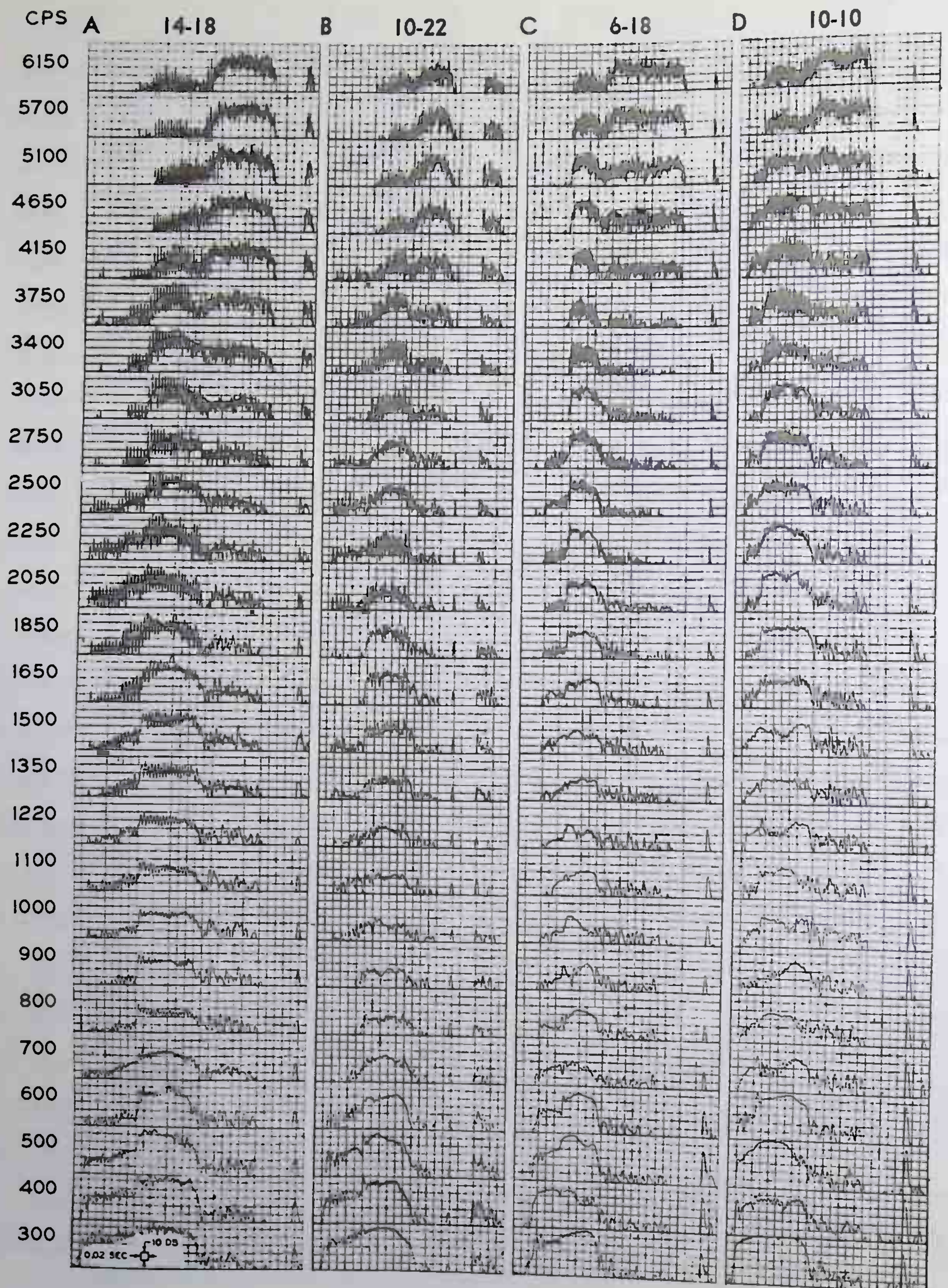
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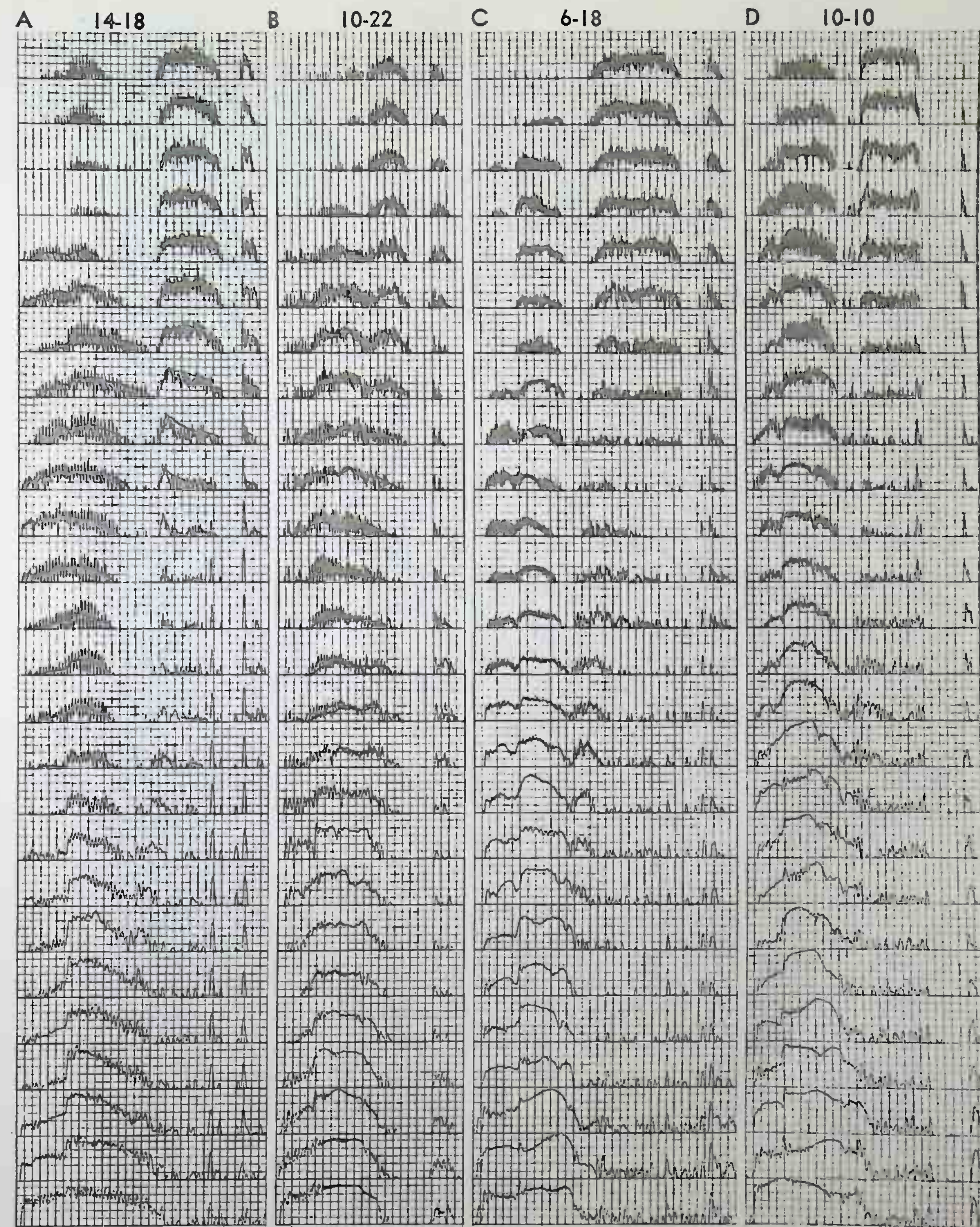
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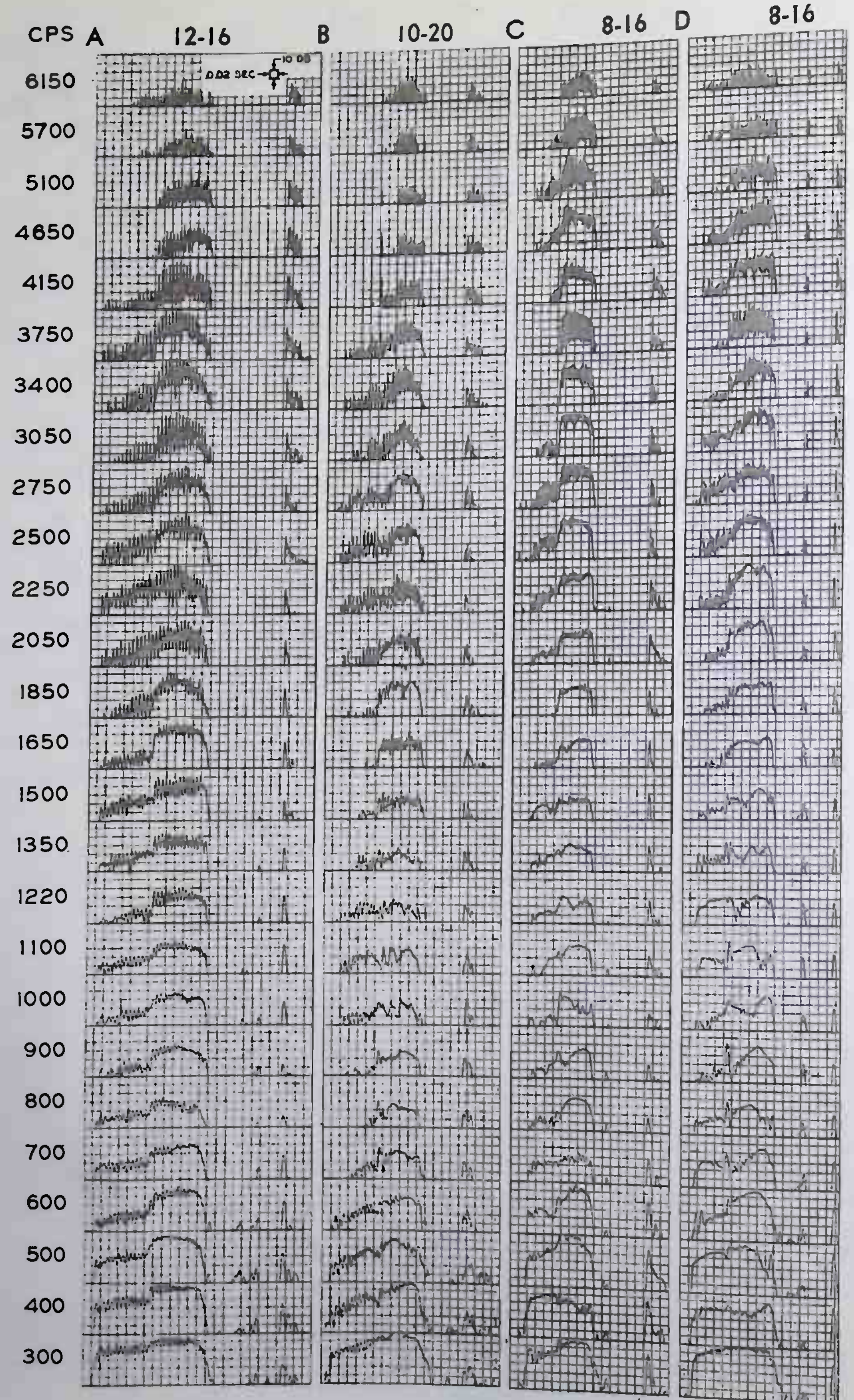

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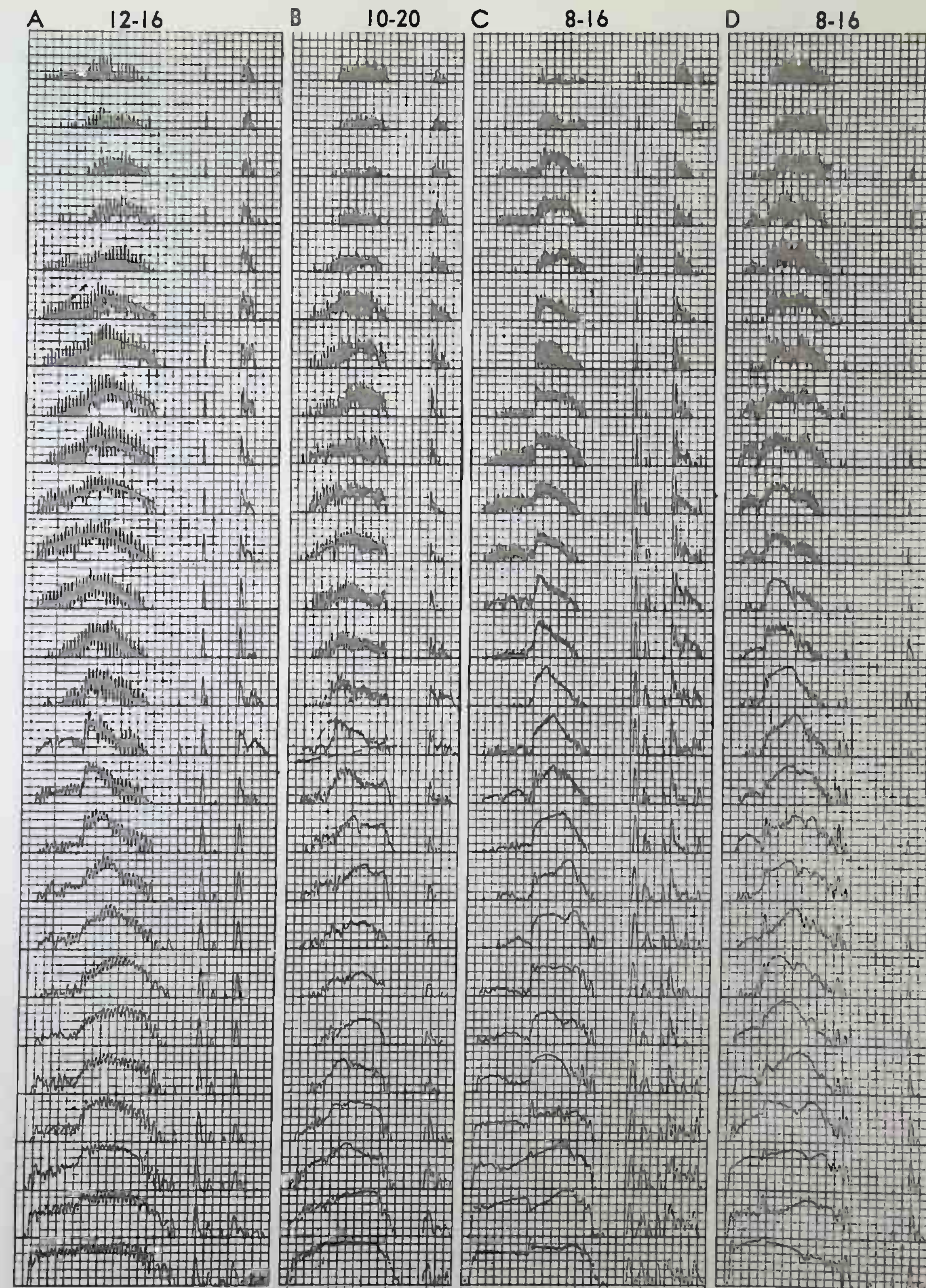

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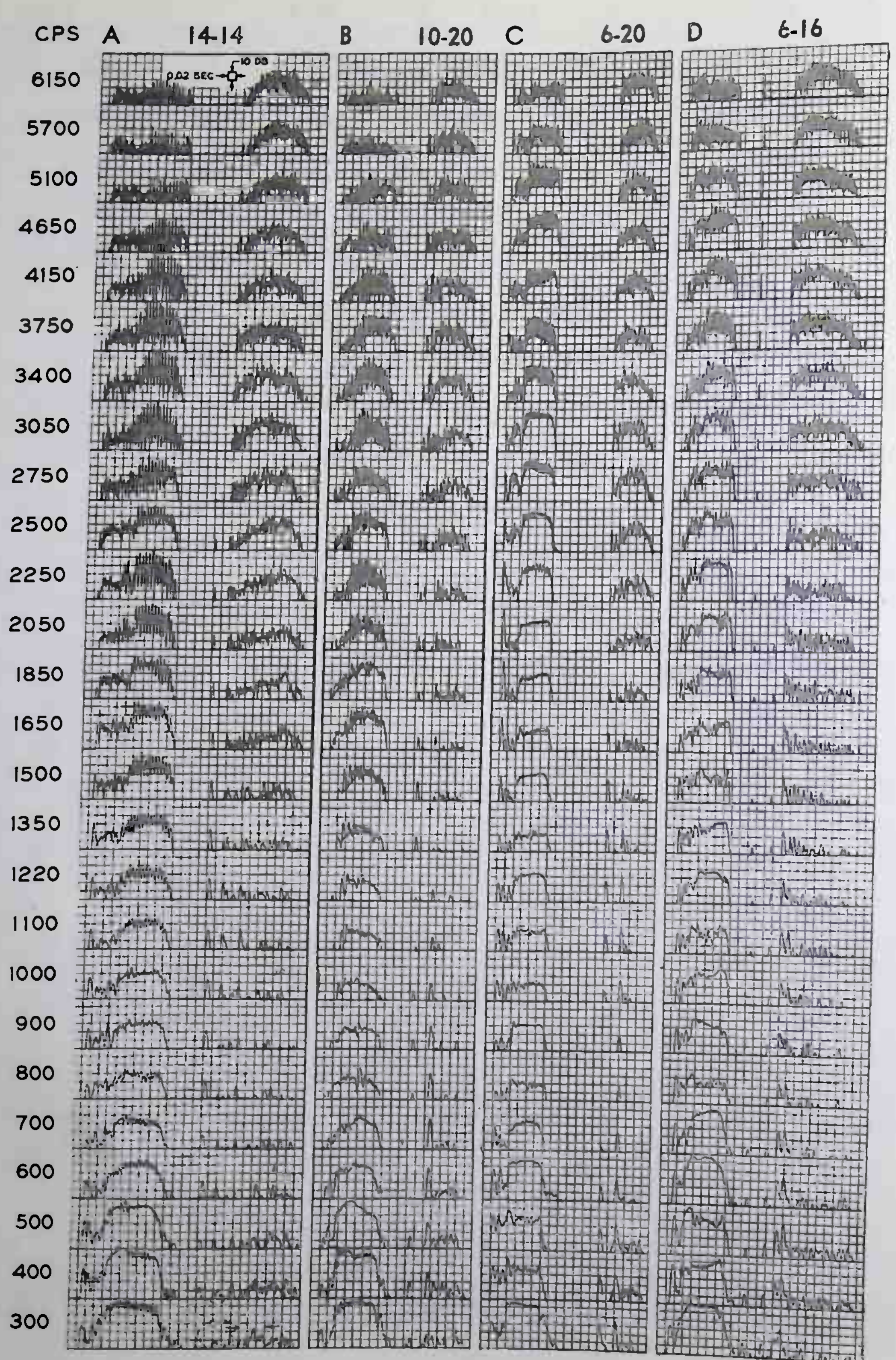

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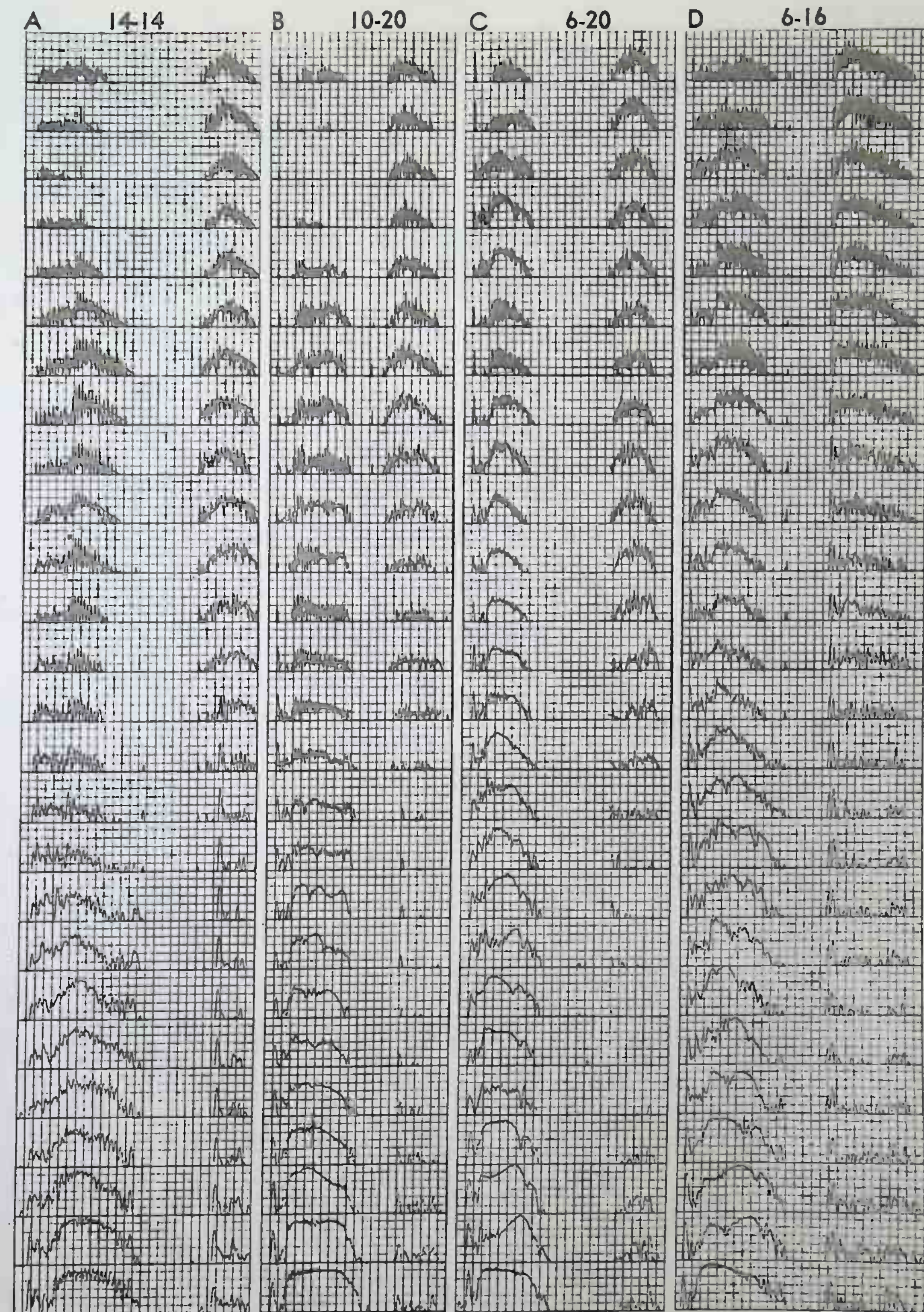

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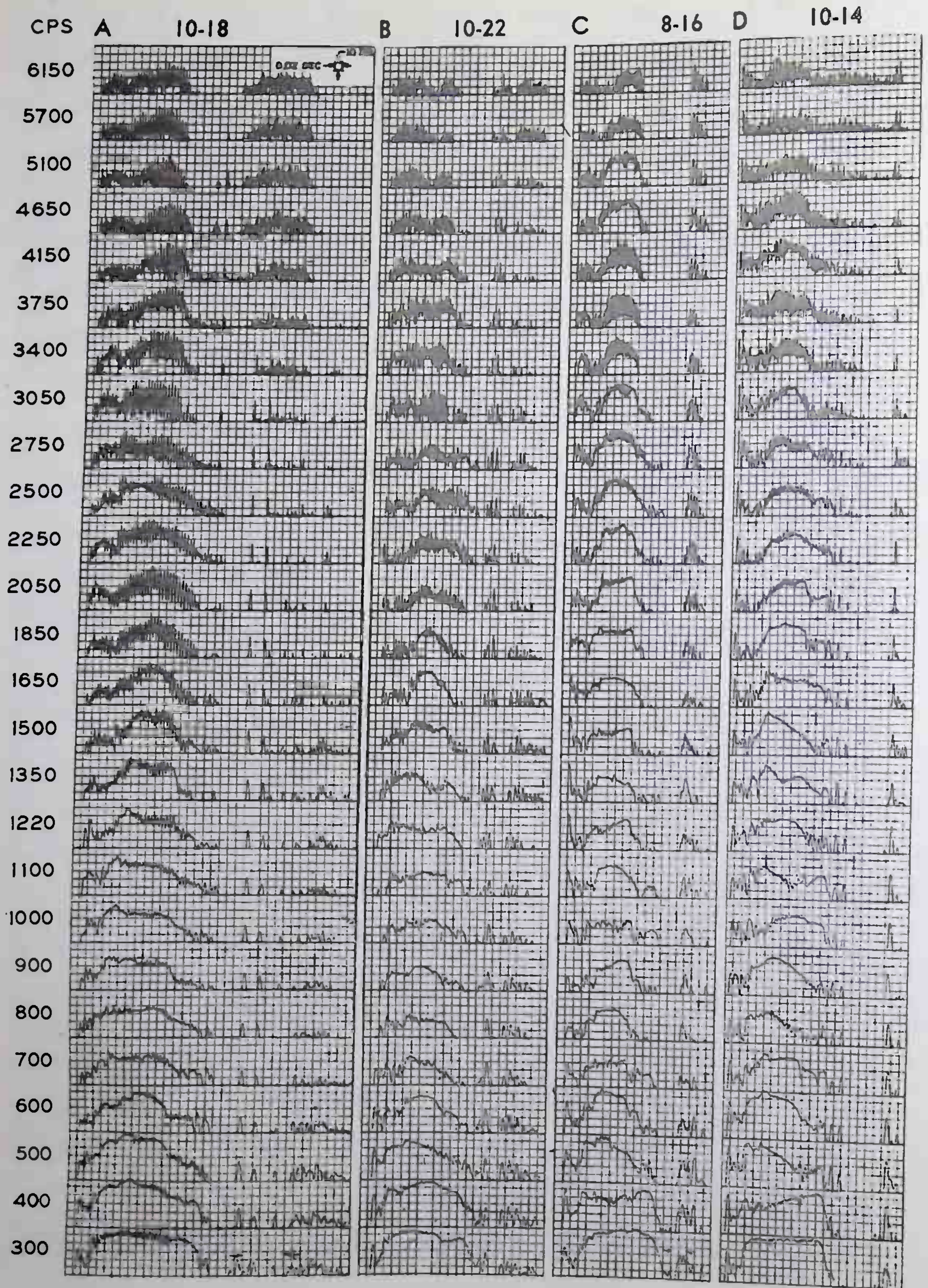

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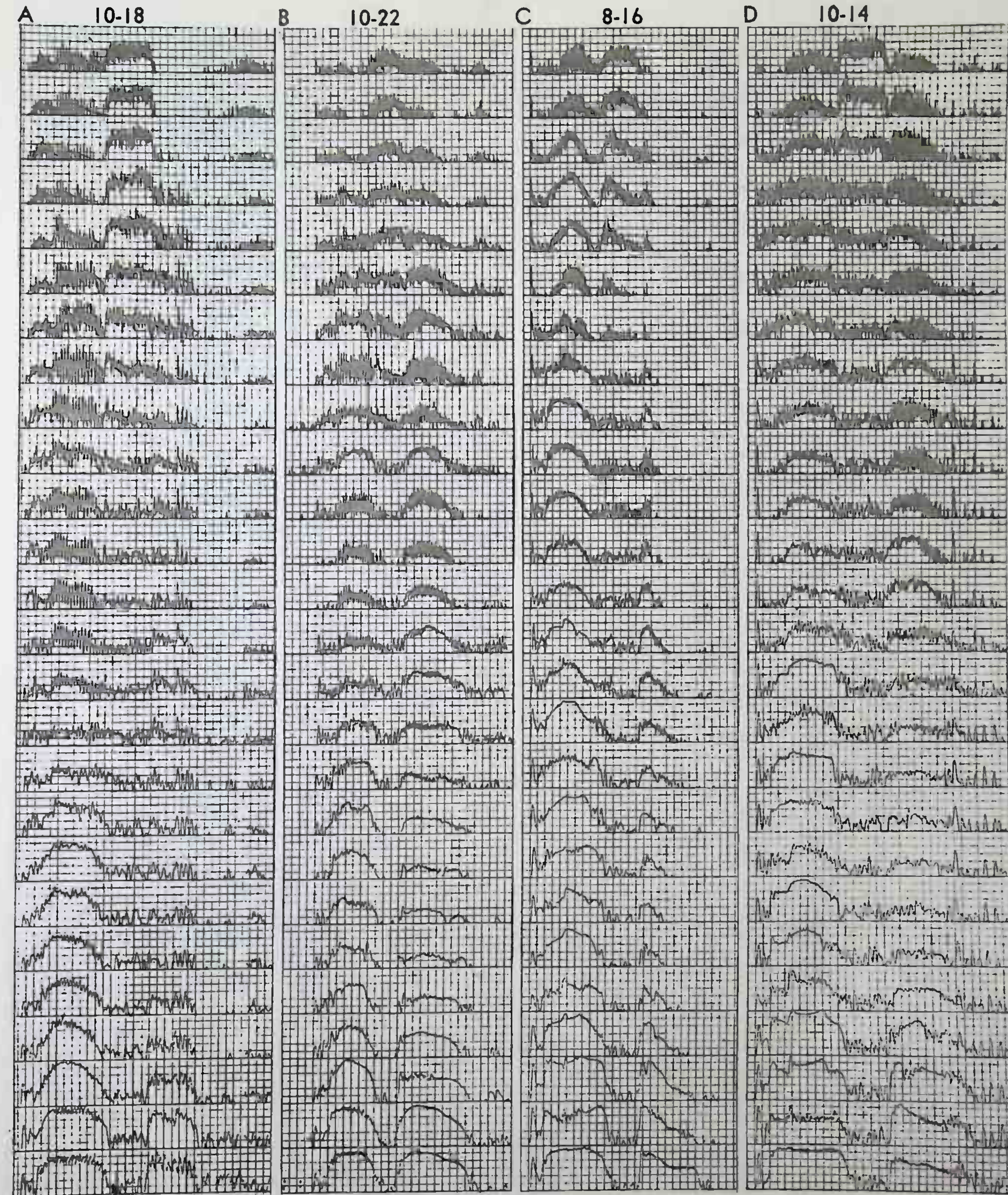

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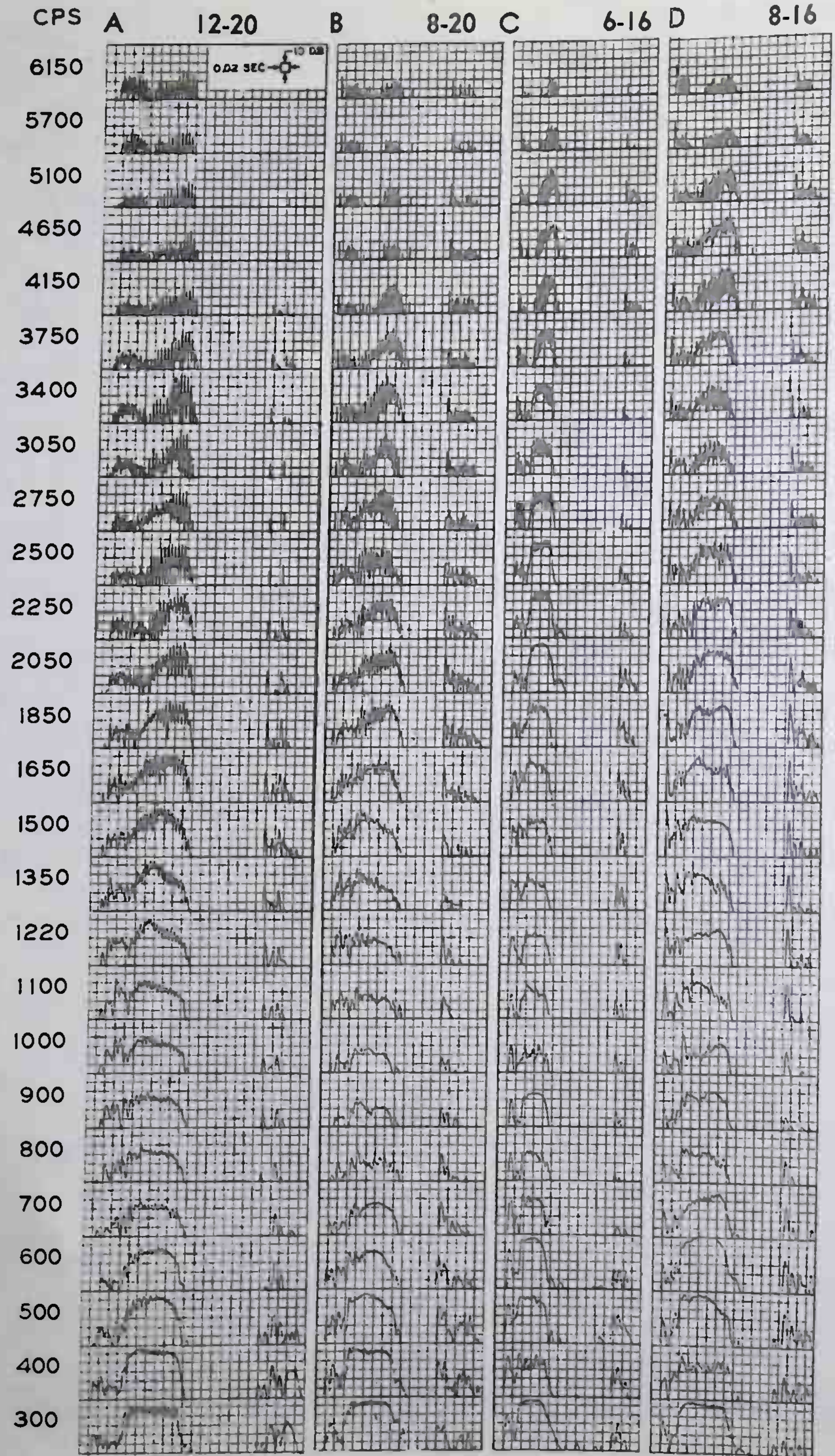

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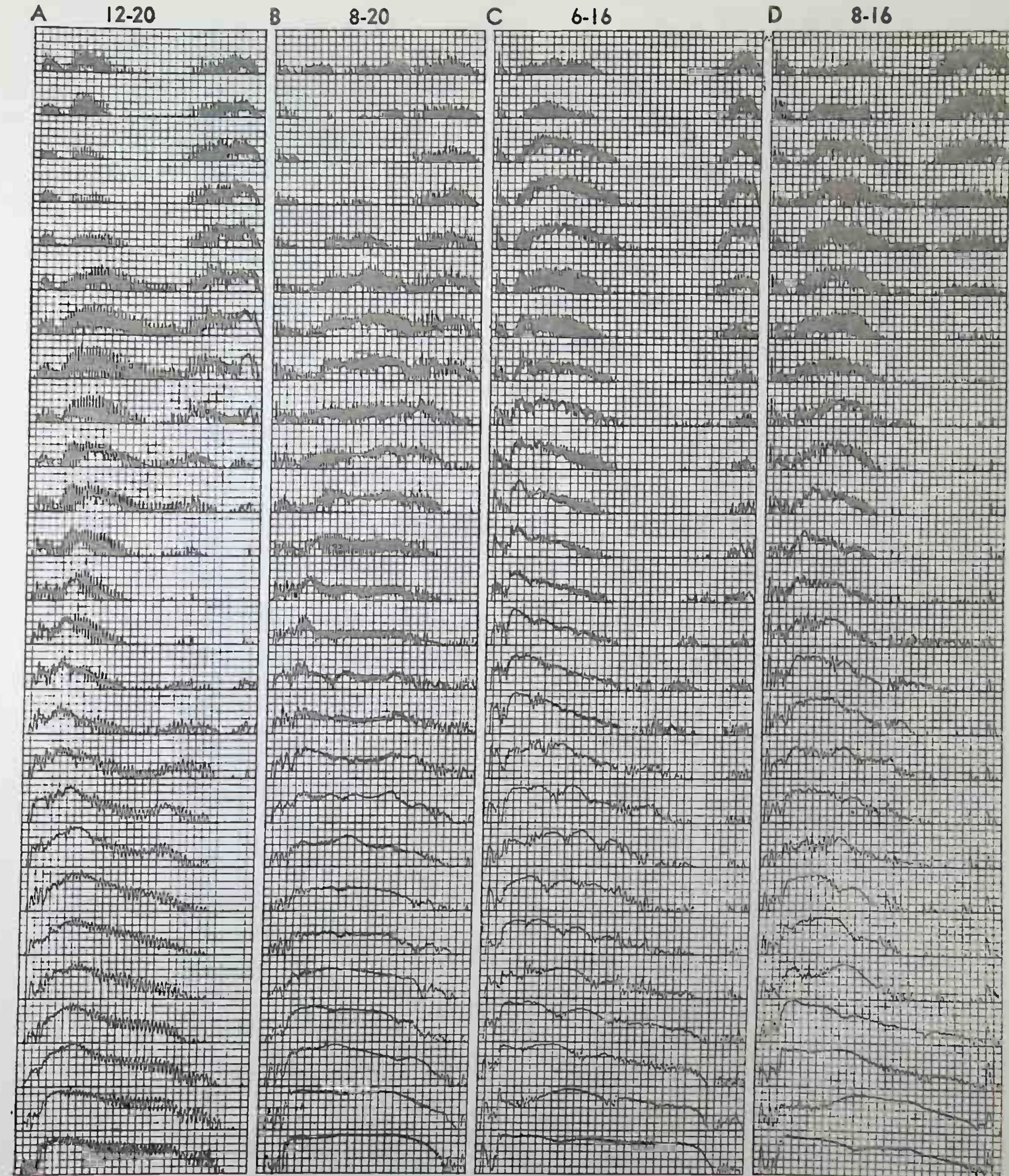
0.02 SEC
10 DB

PROSE



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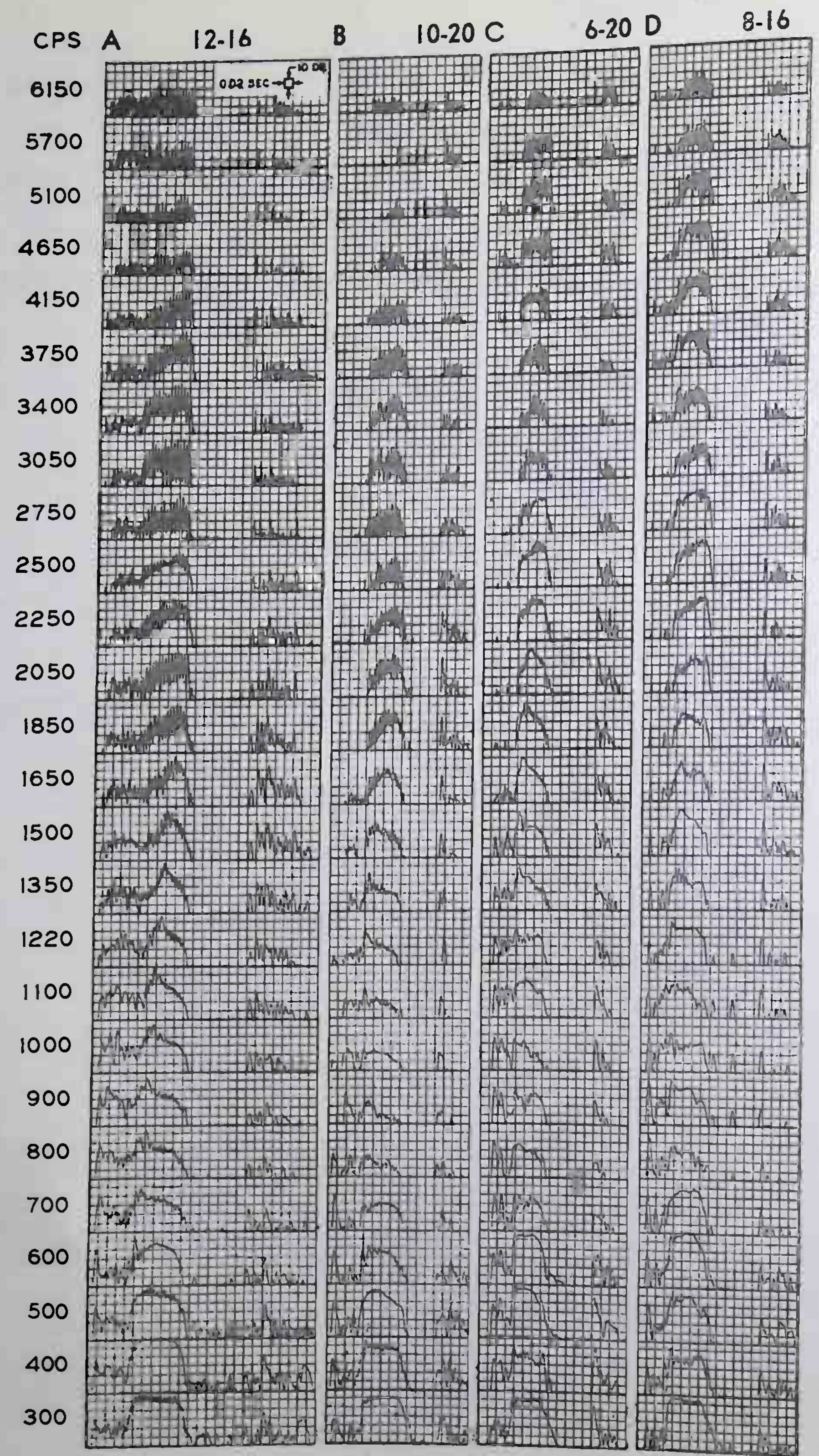
QUICK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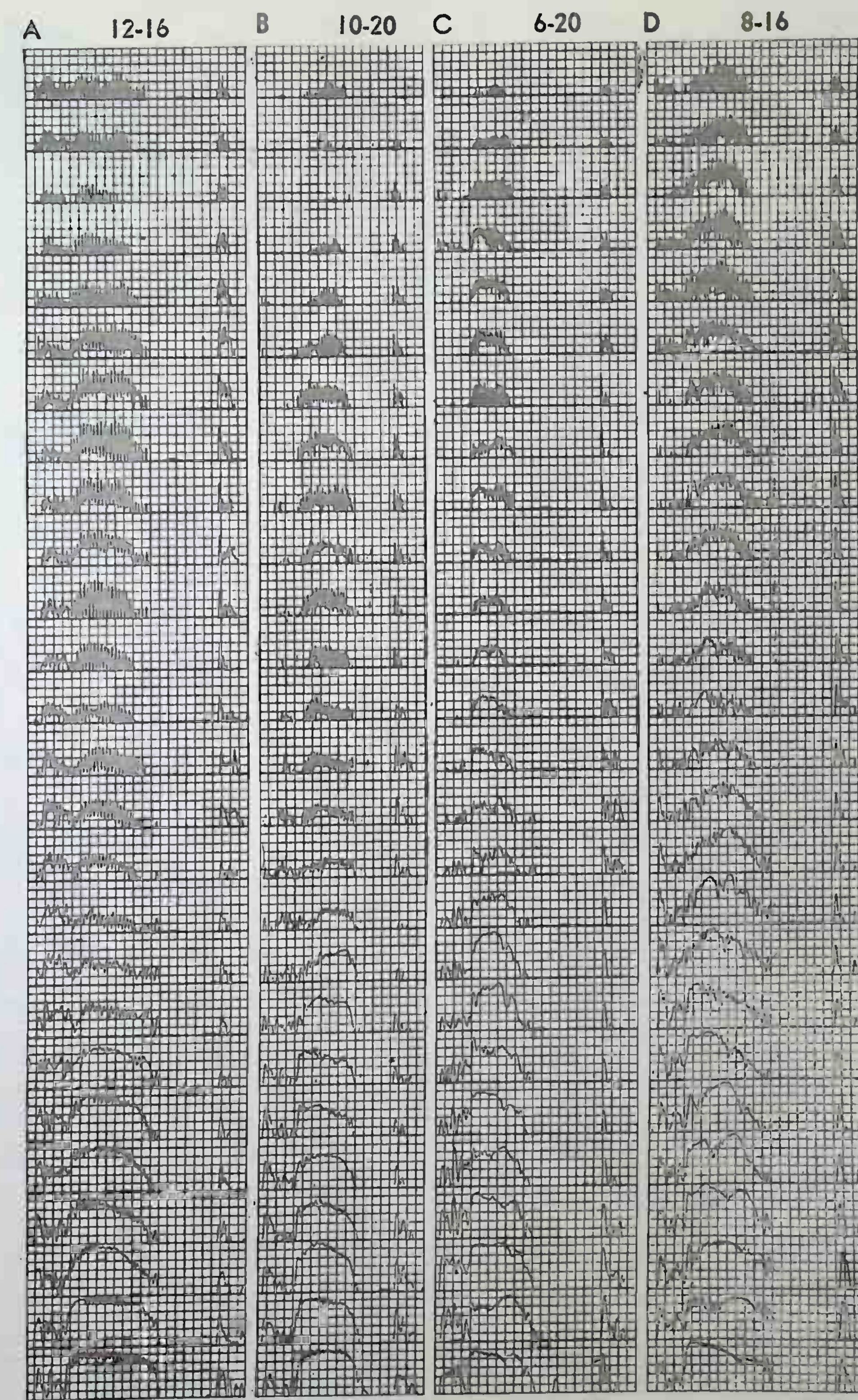
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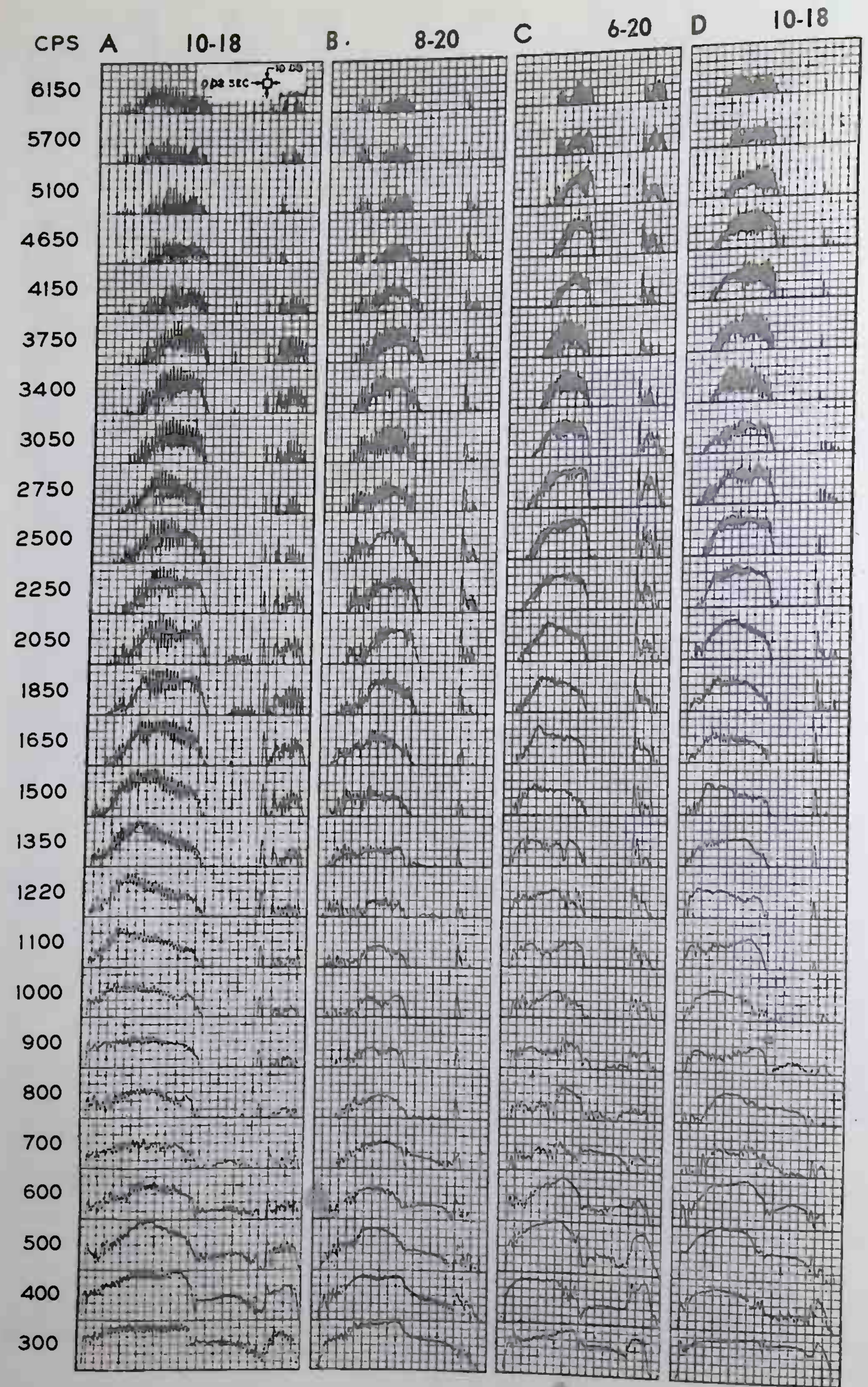
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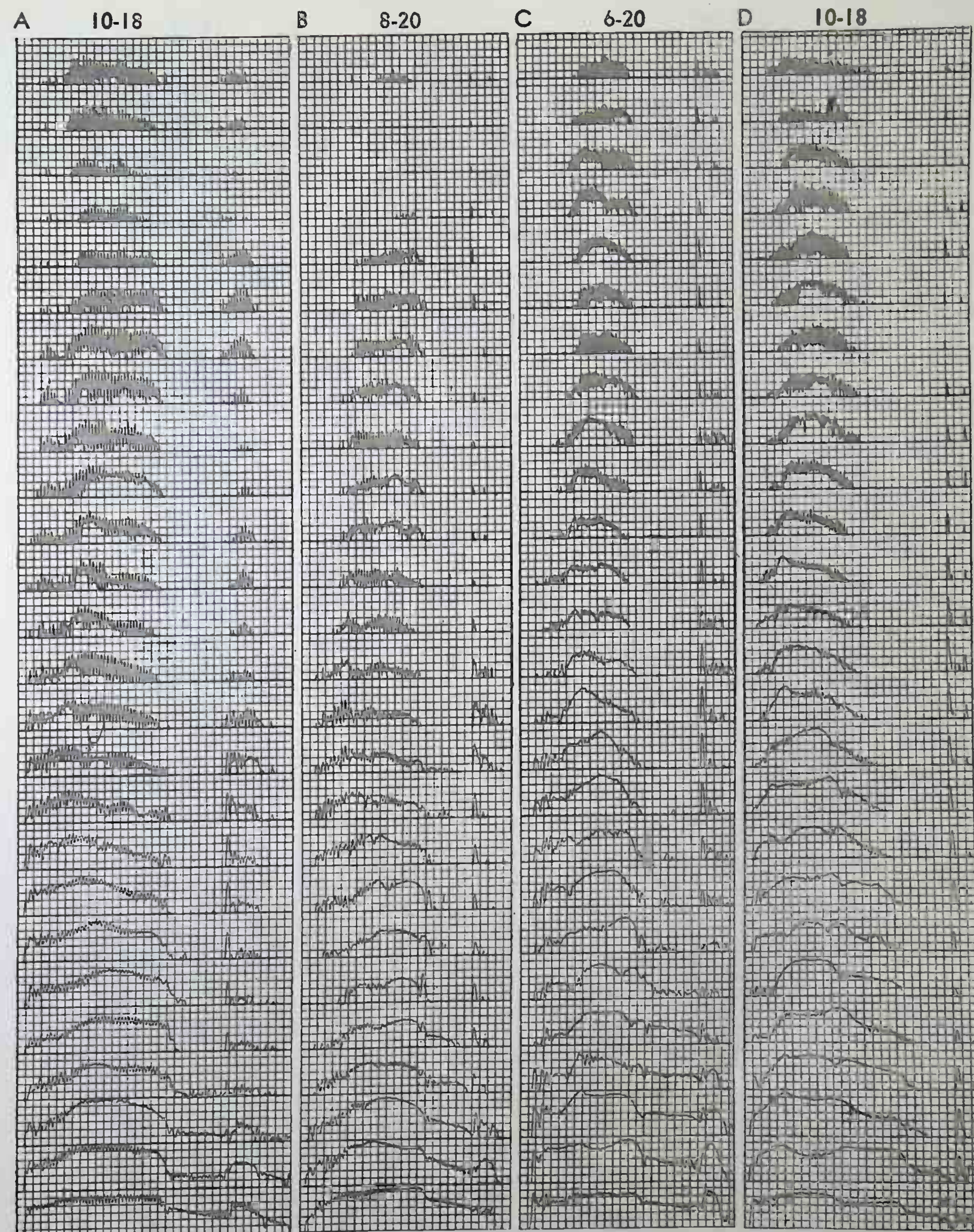
ROGUE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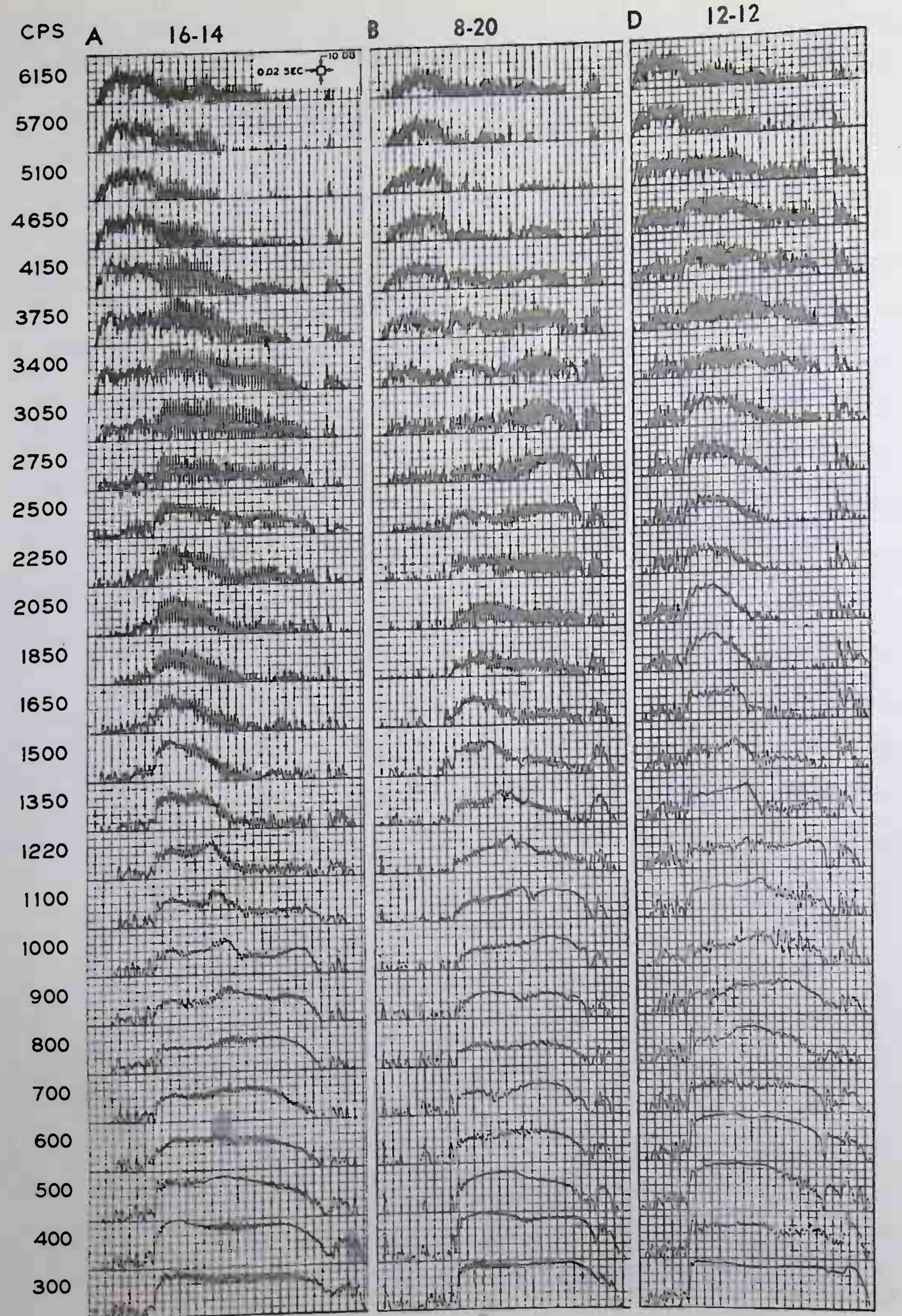
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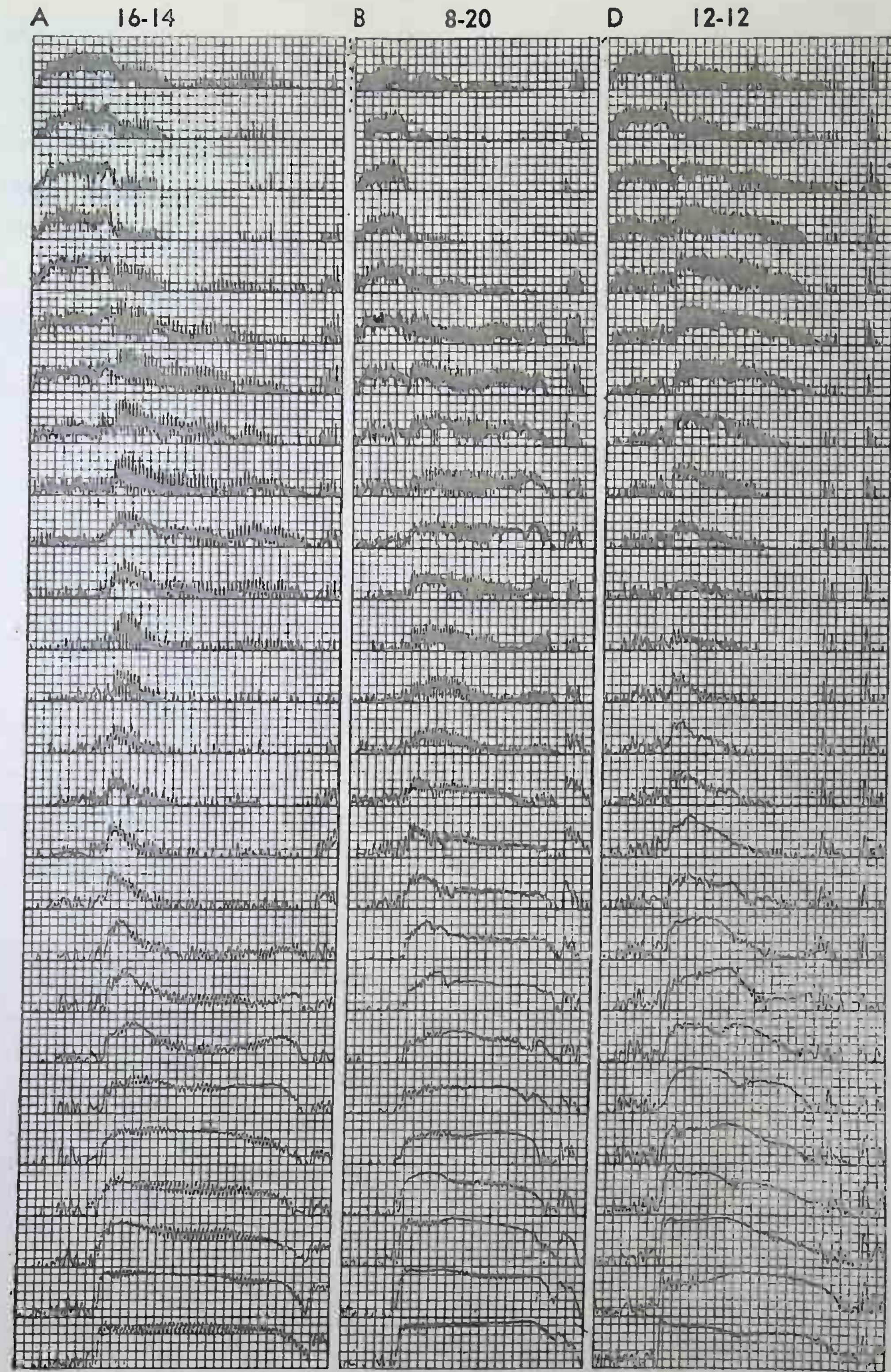
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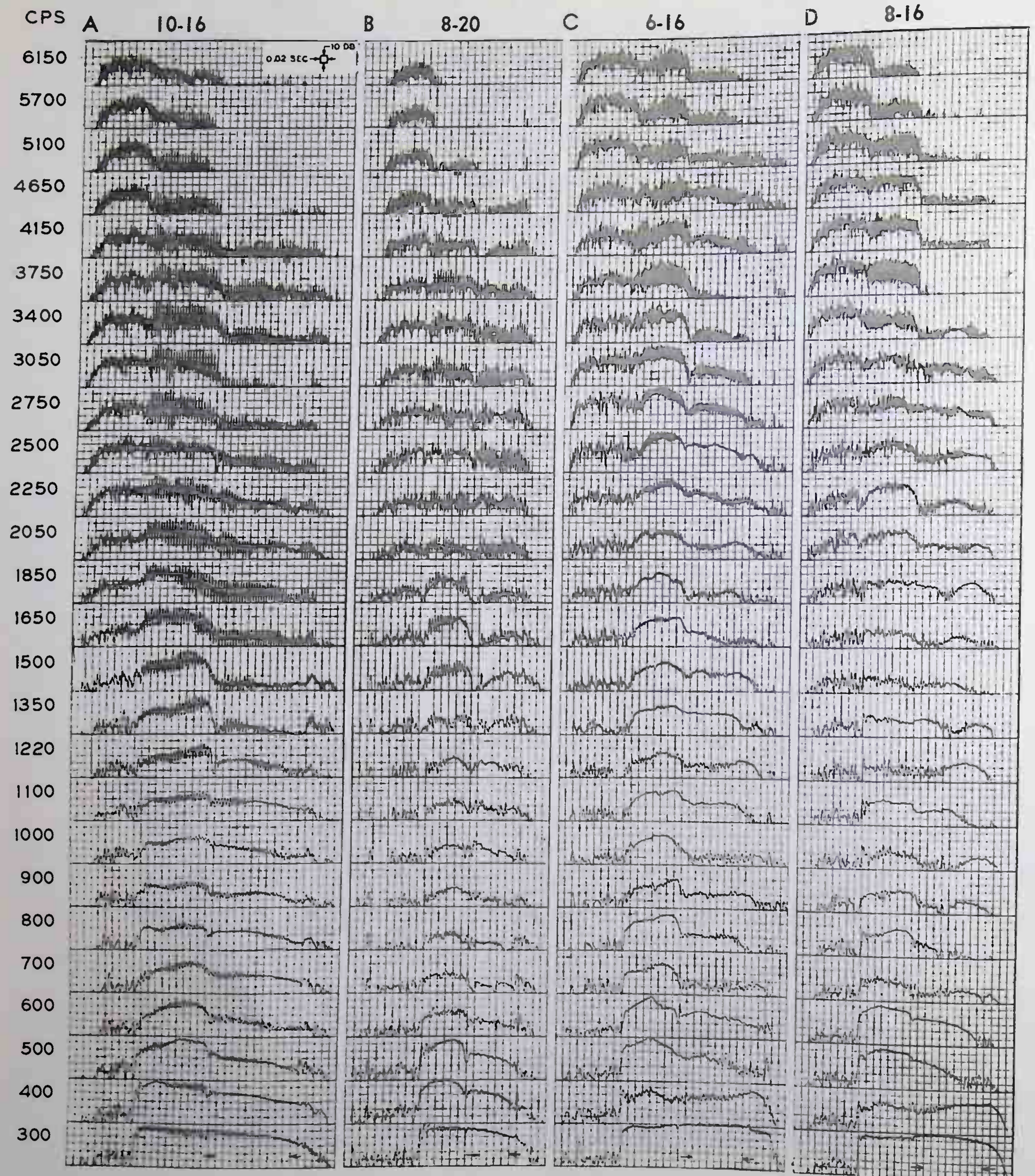
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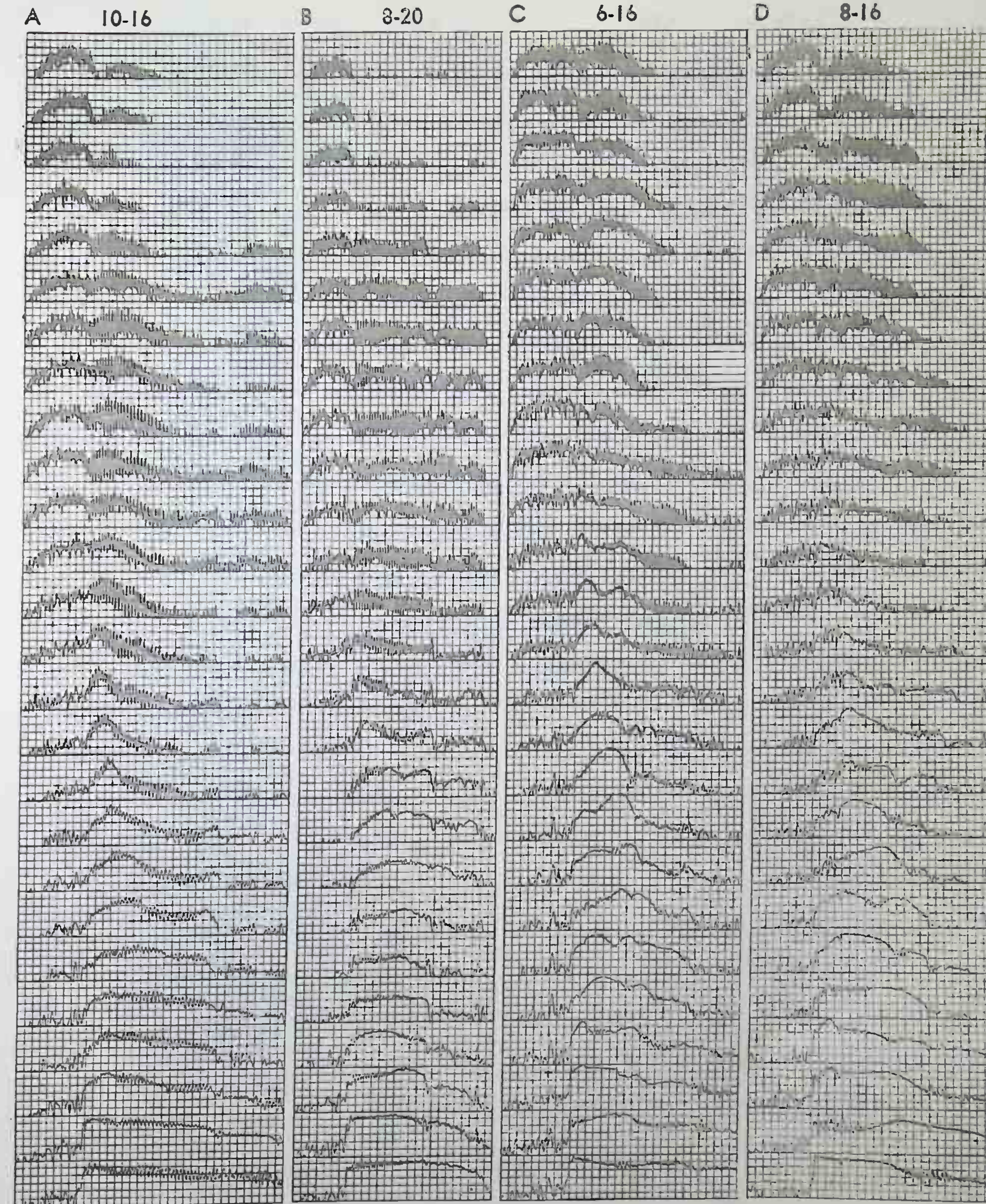
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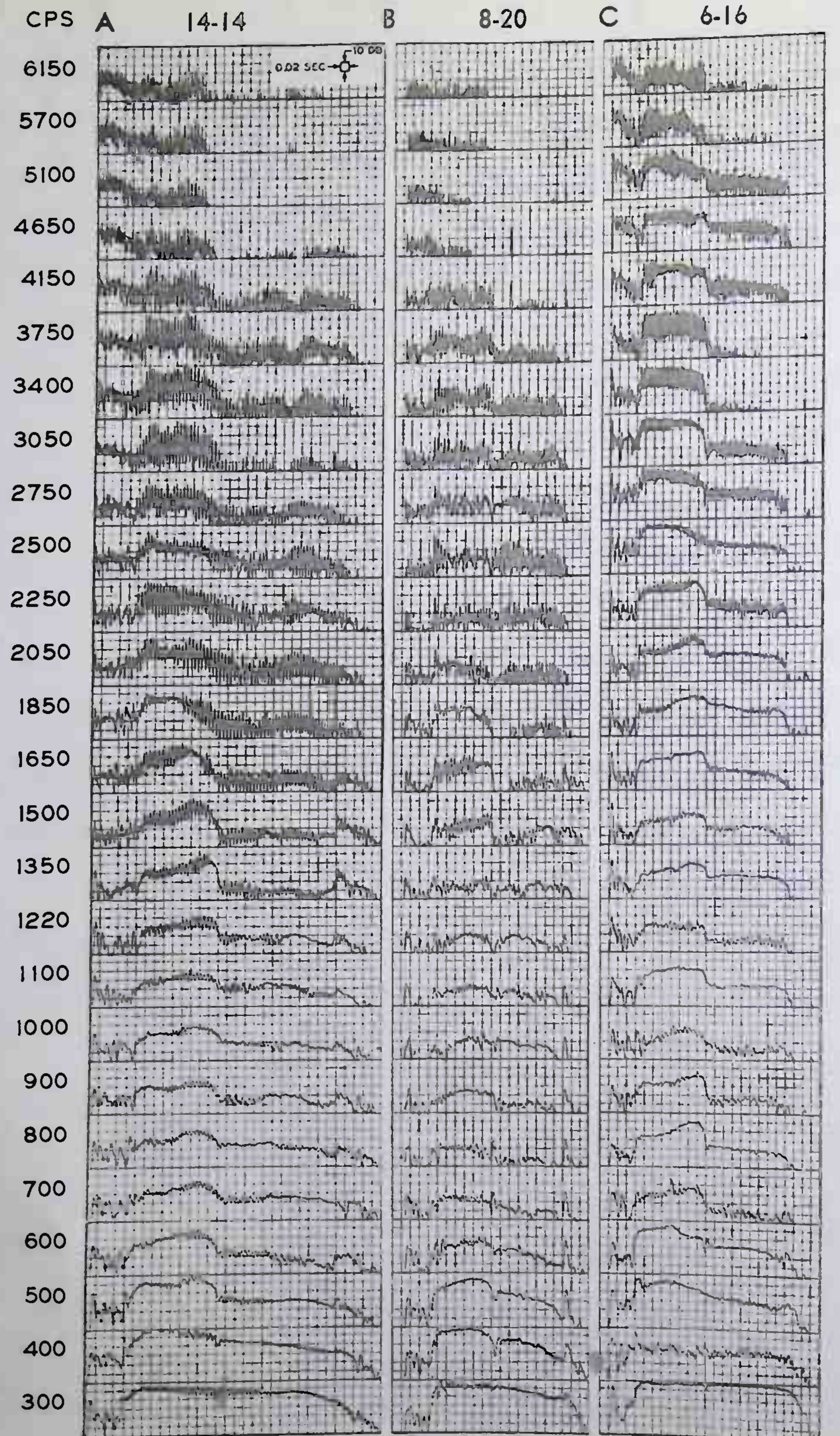
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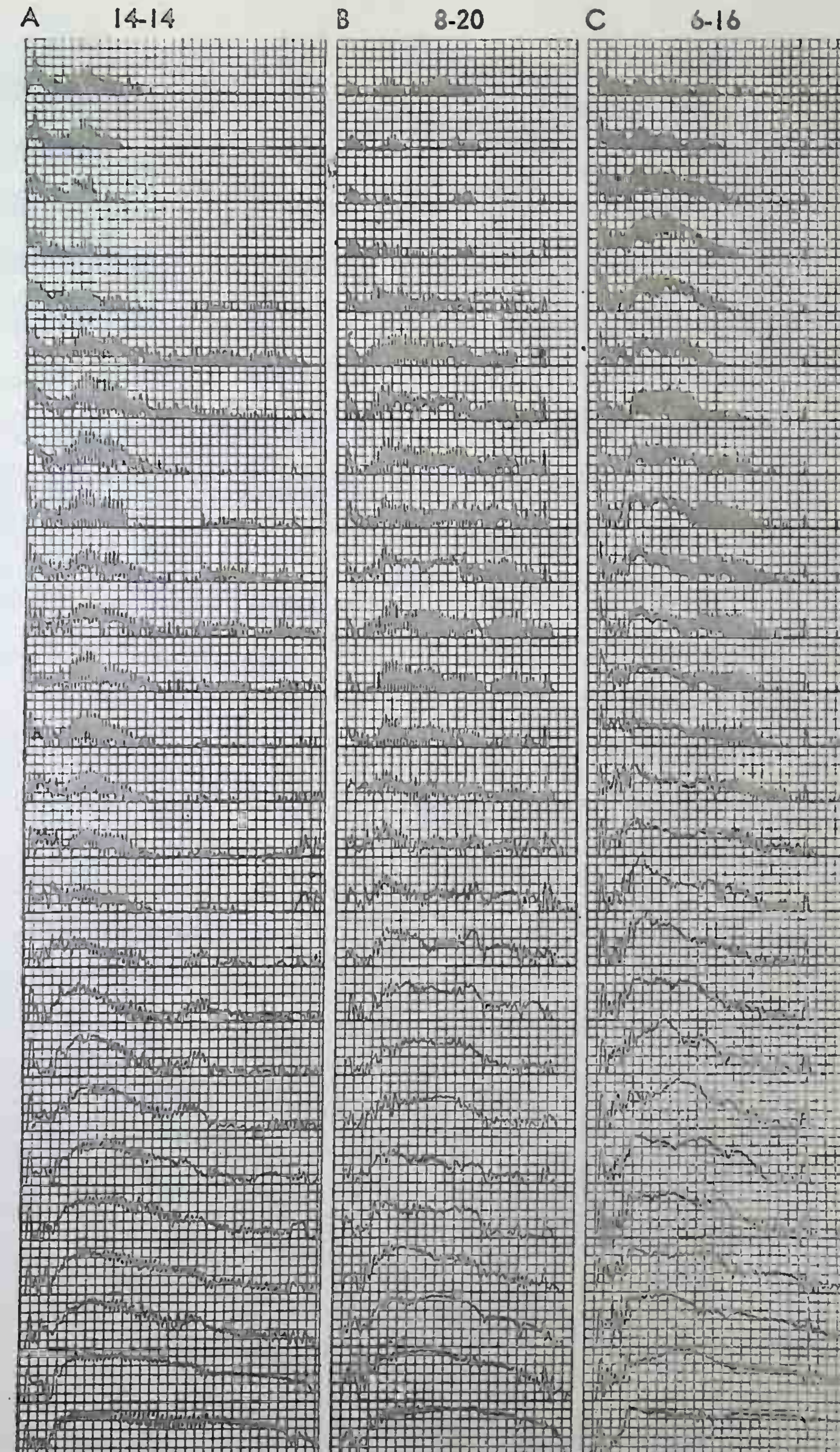
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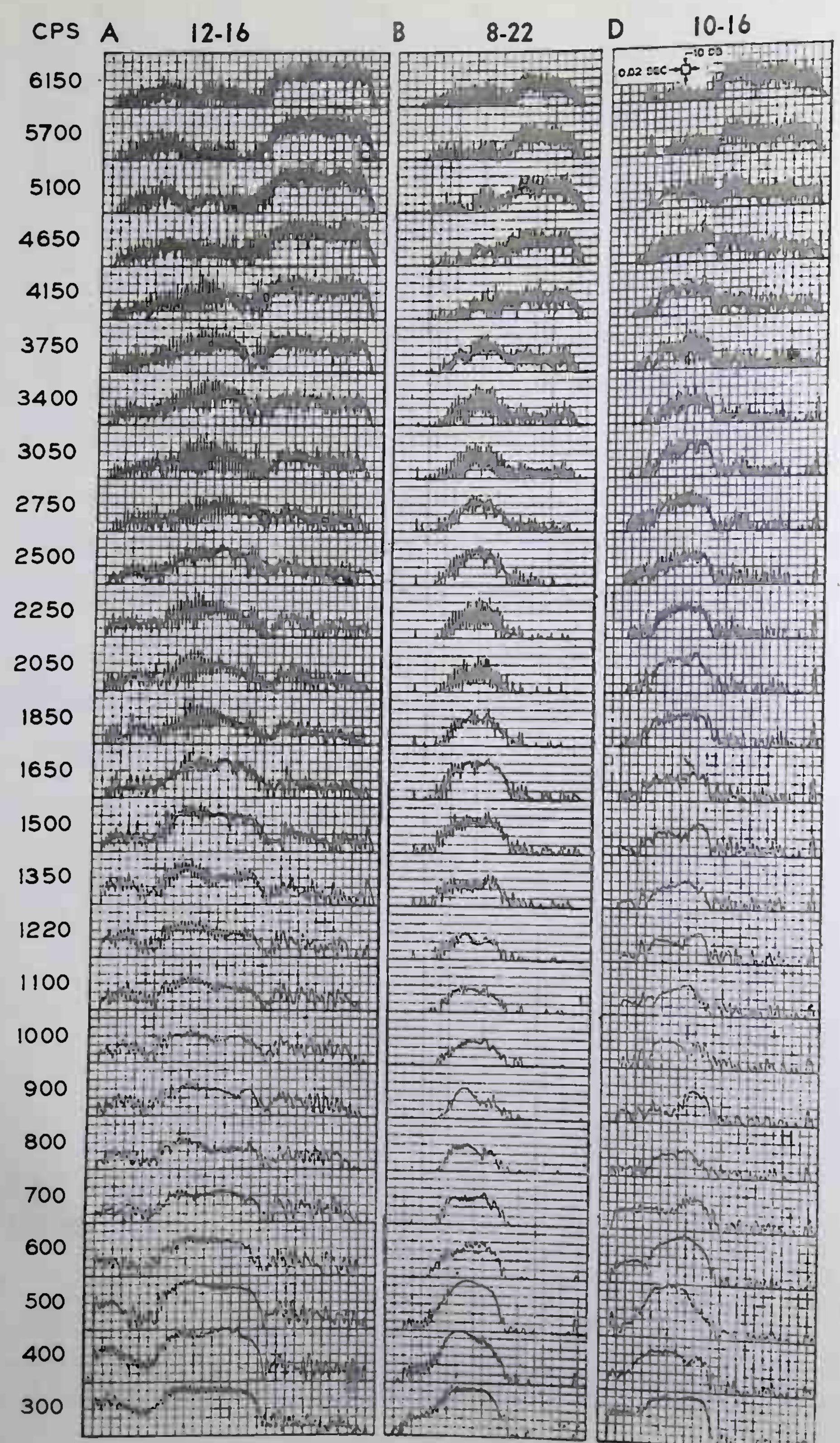
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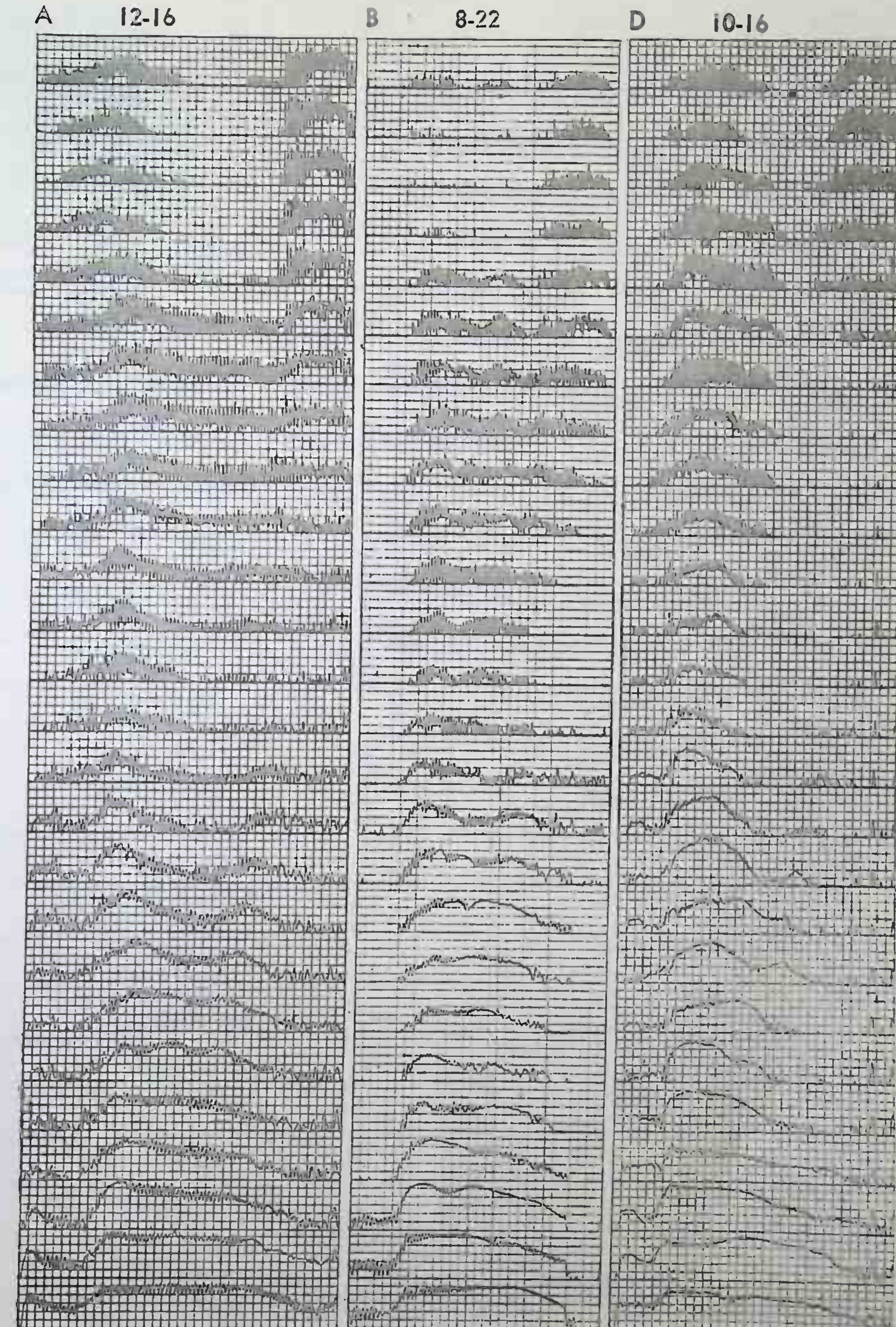
THOSE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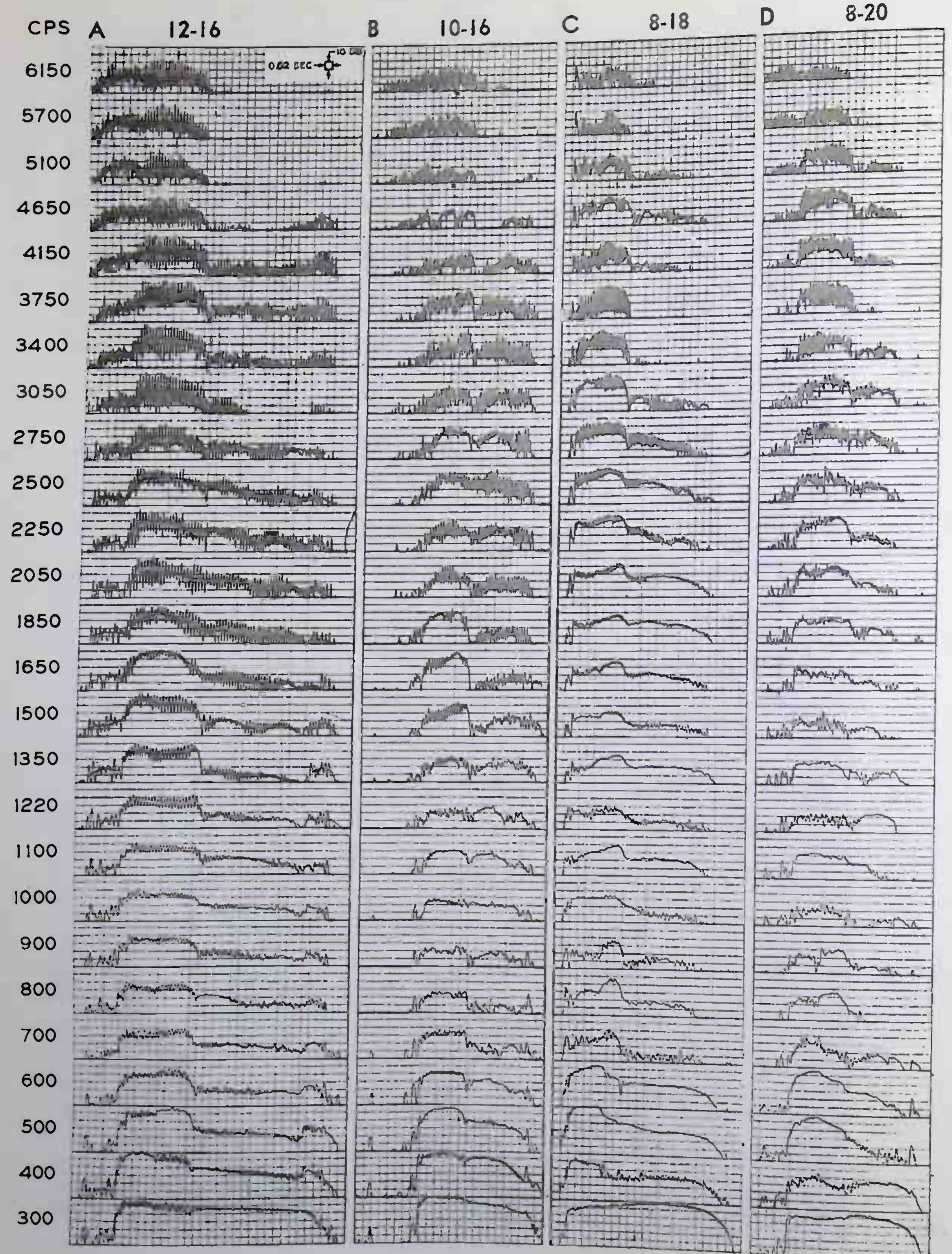
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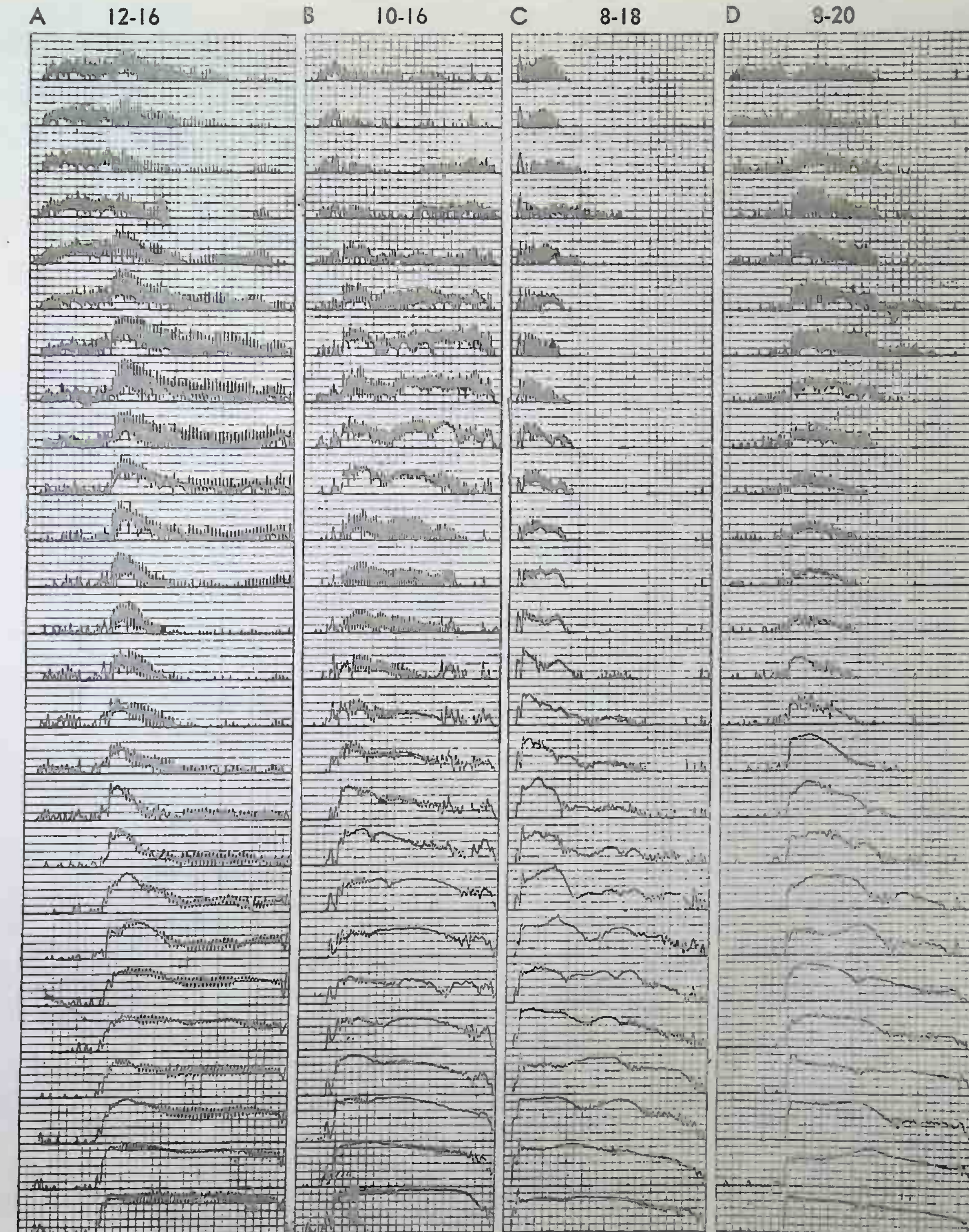
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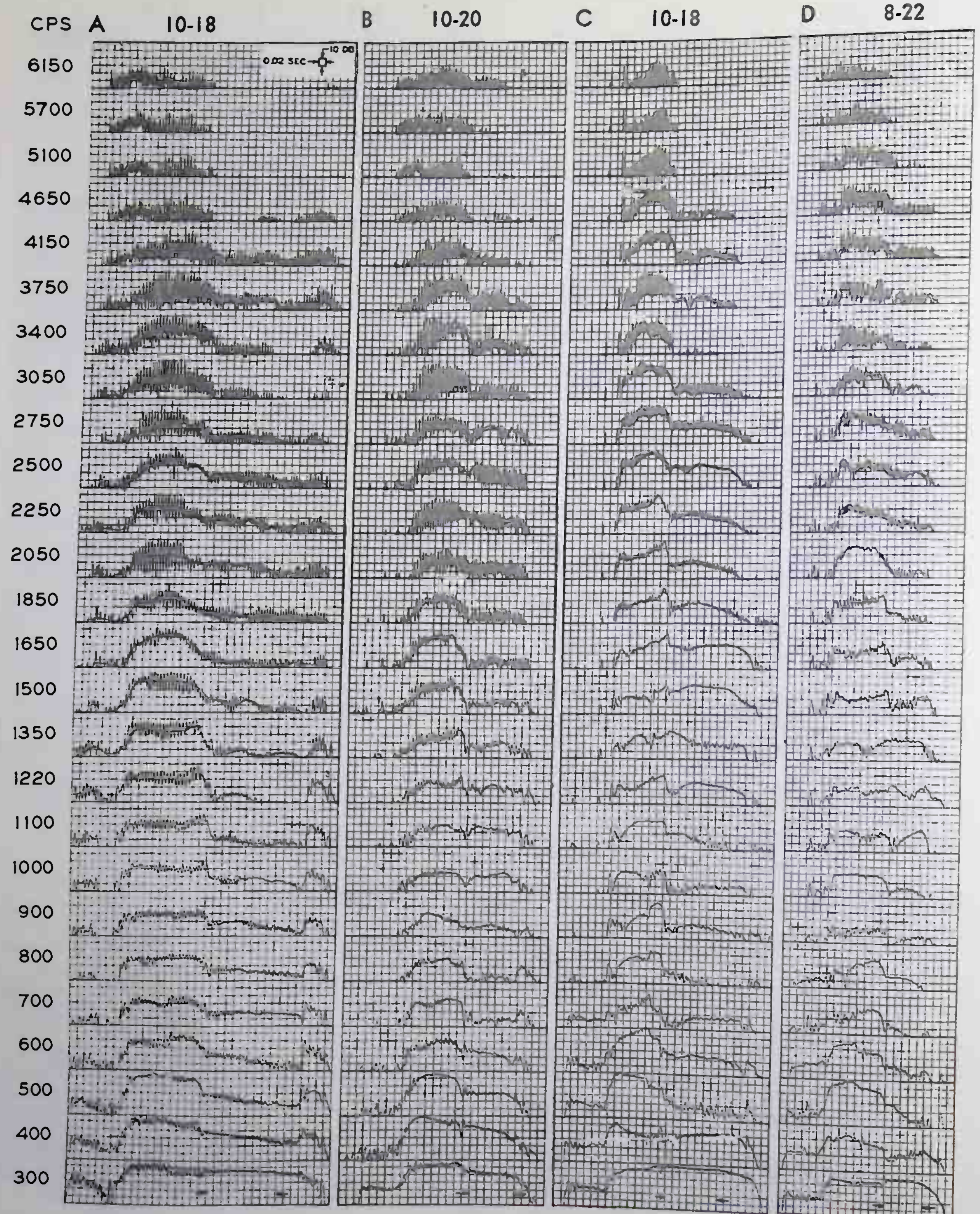

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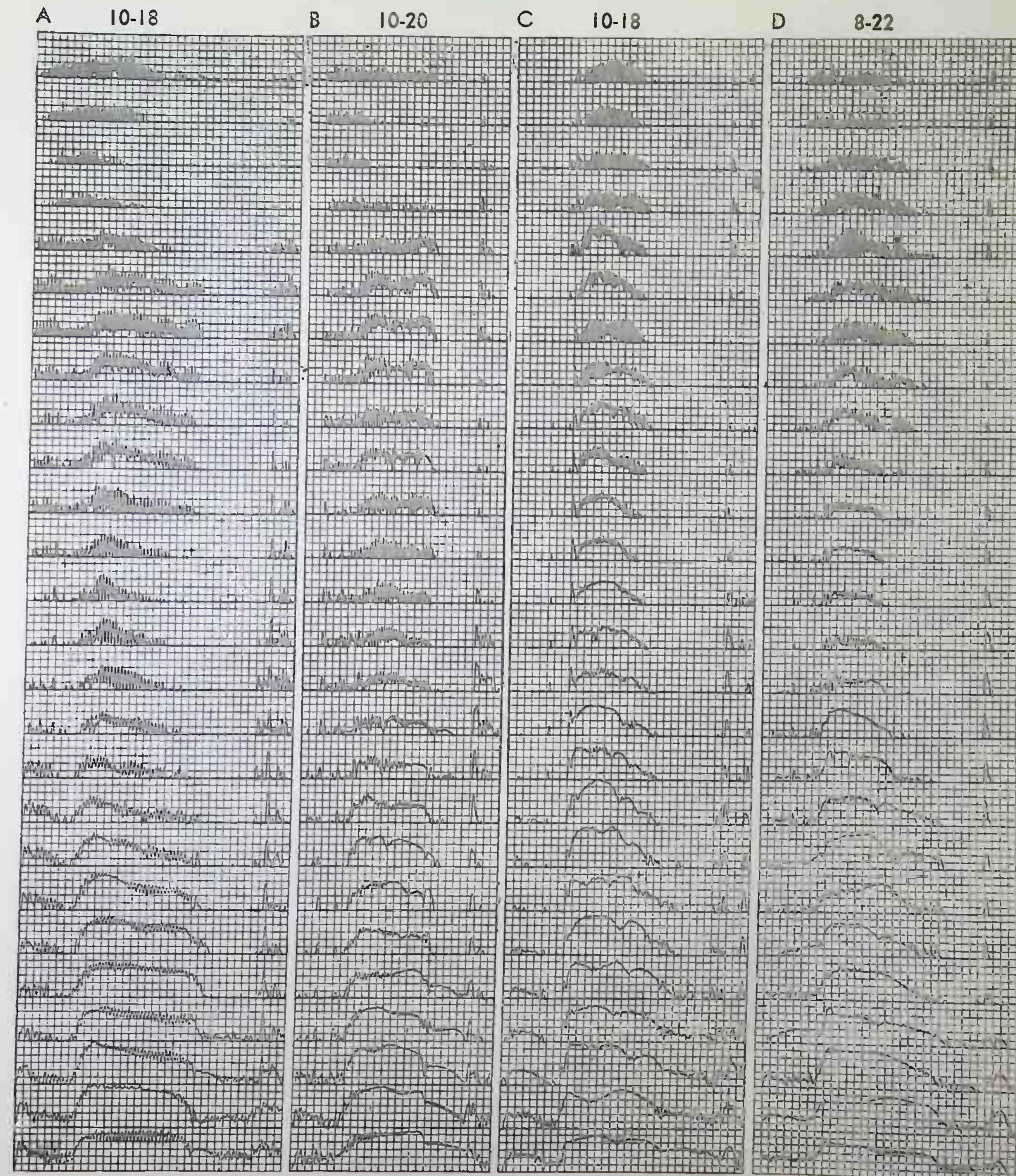
VOGUE



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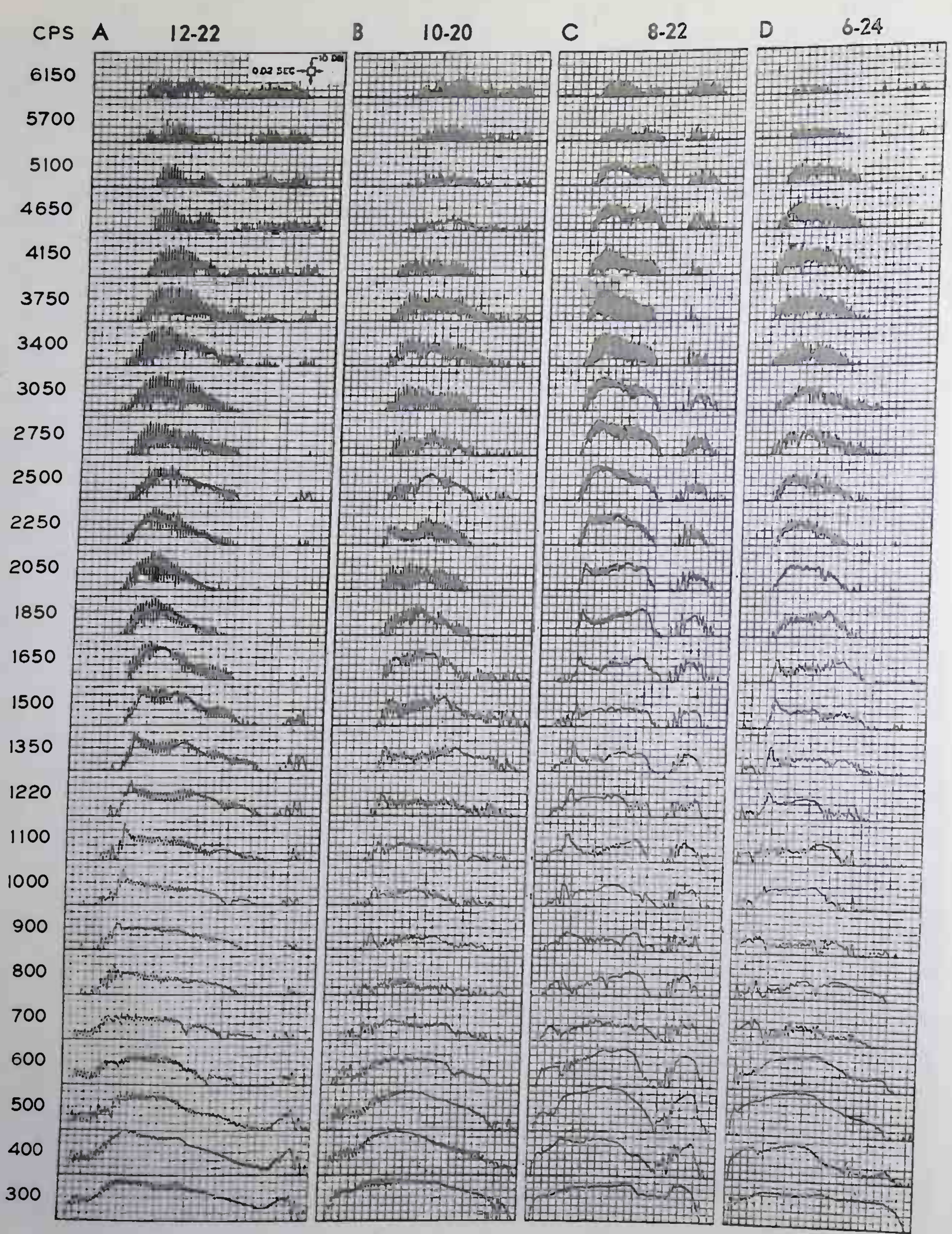
WITH (soft)



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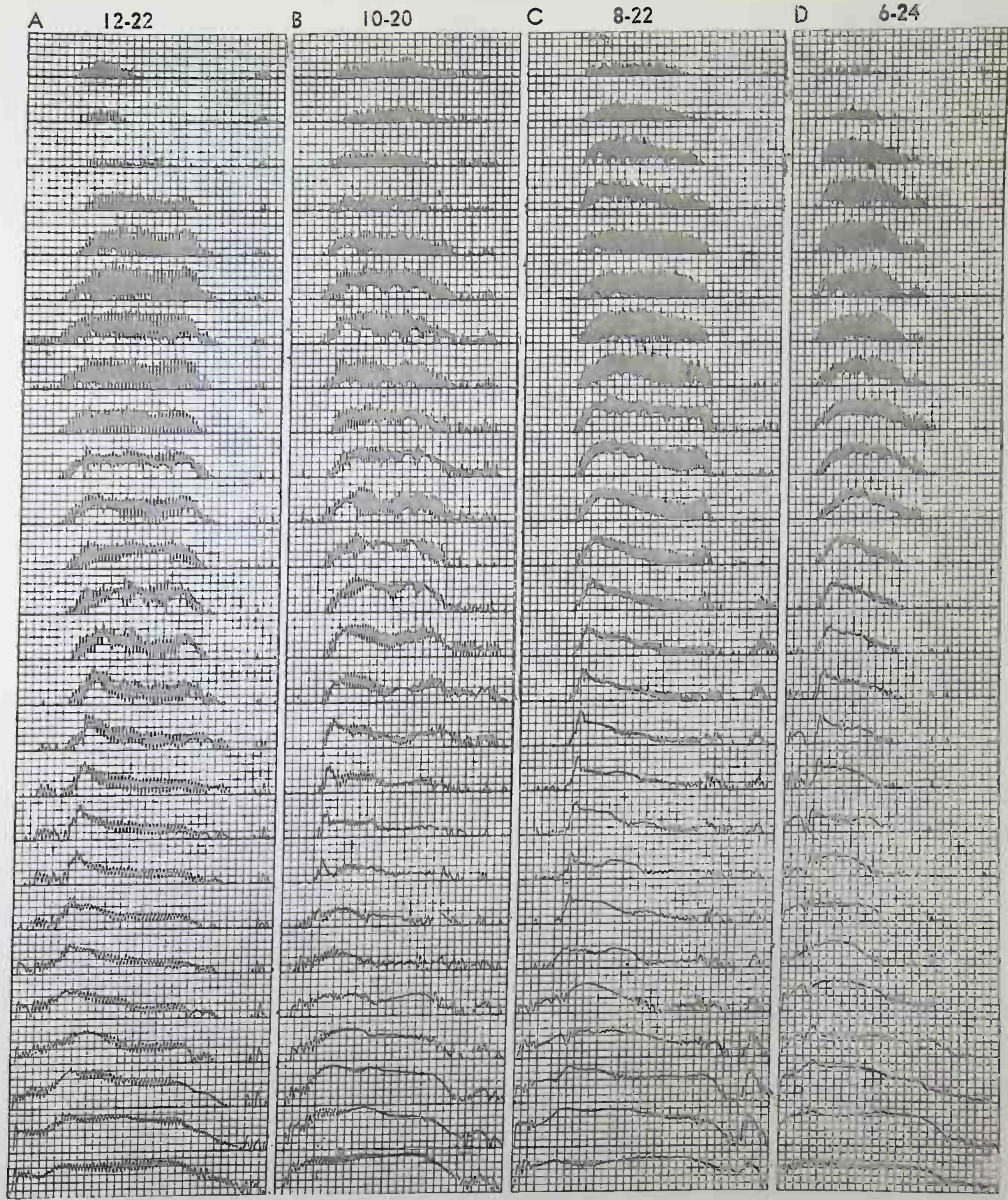

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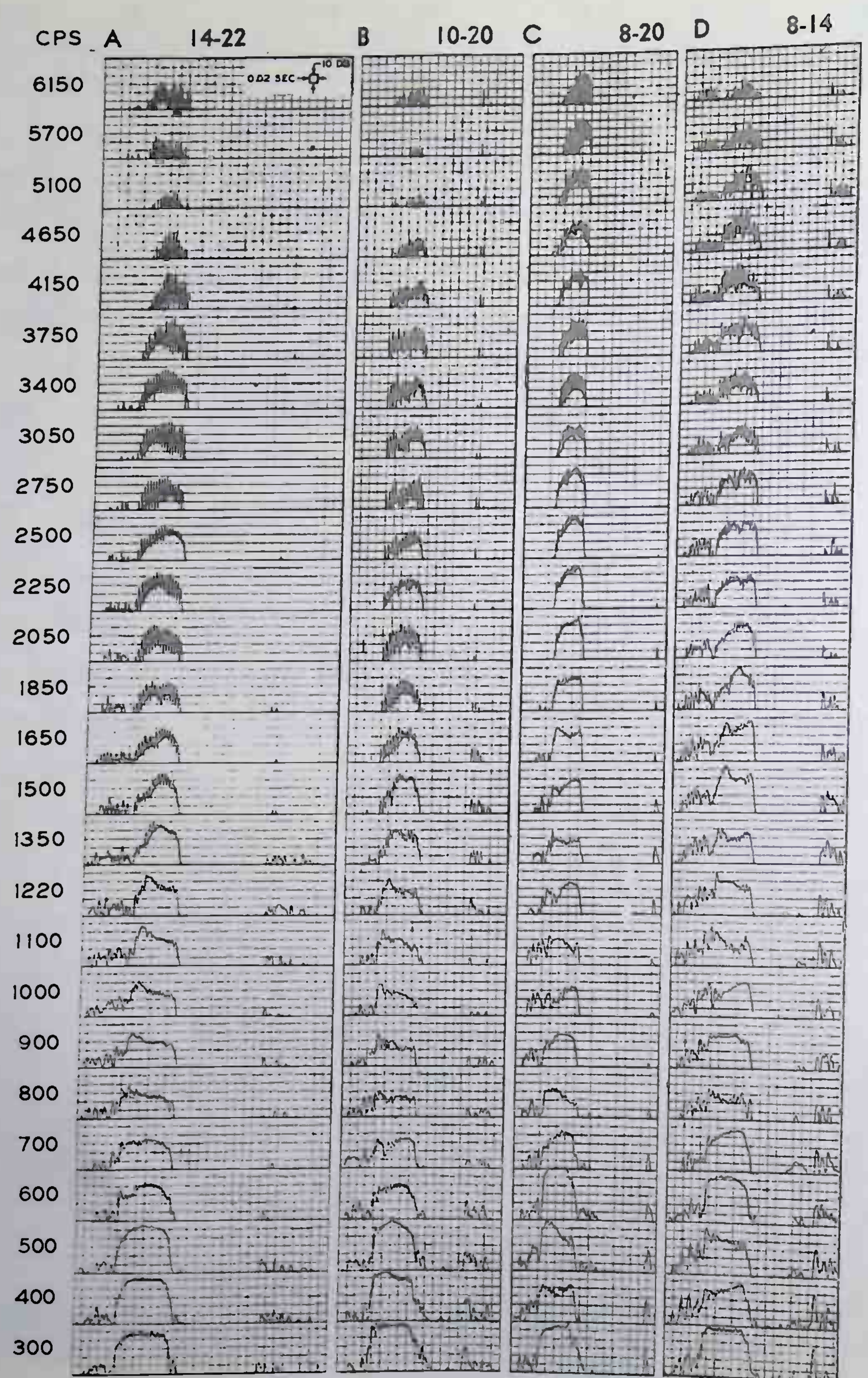
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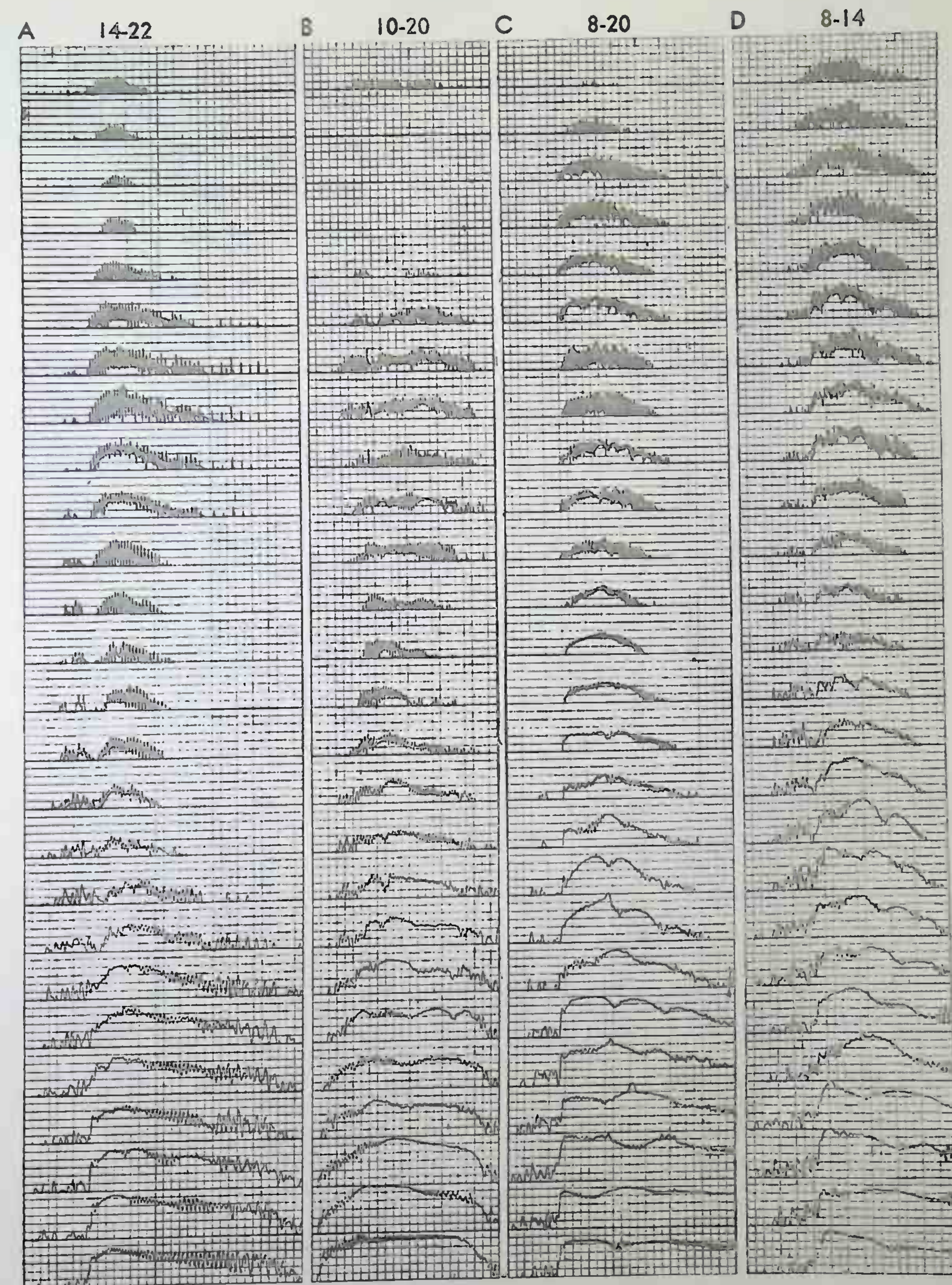
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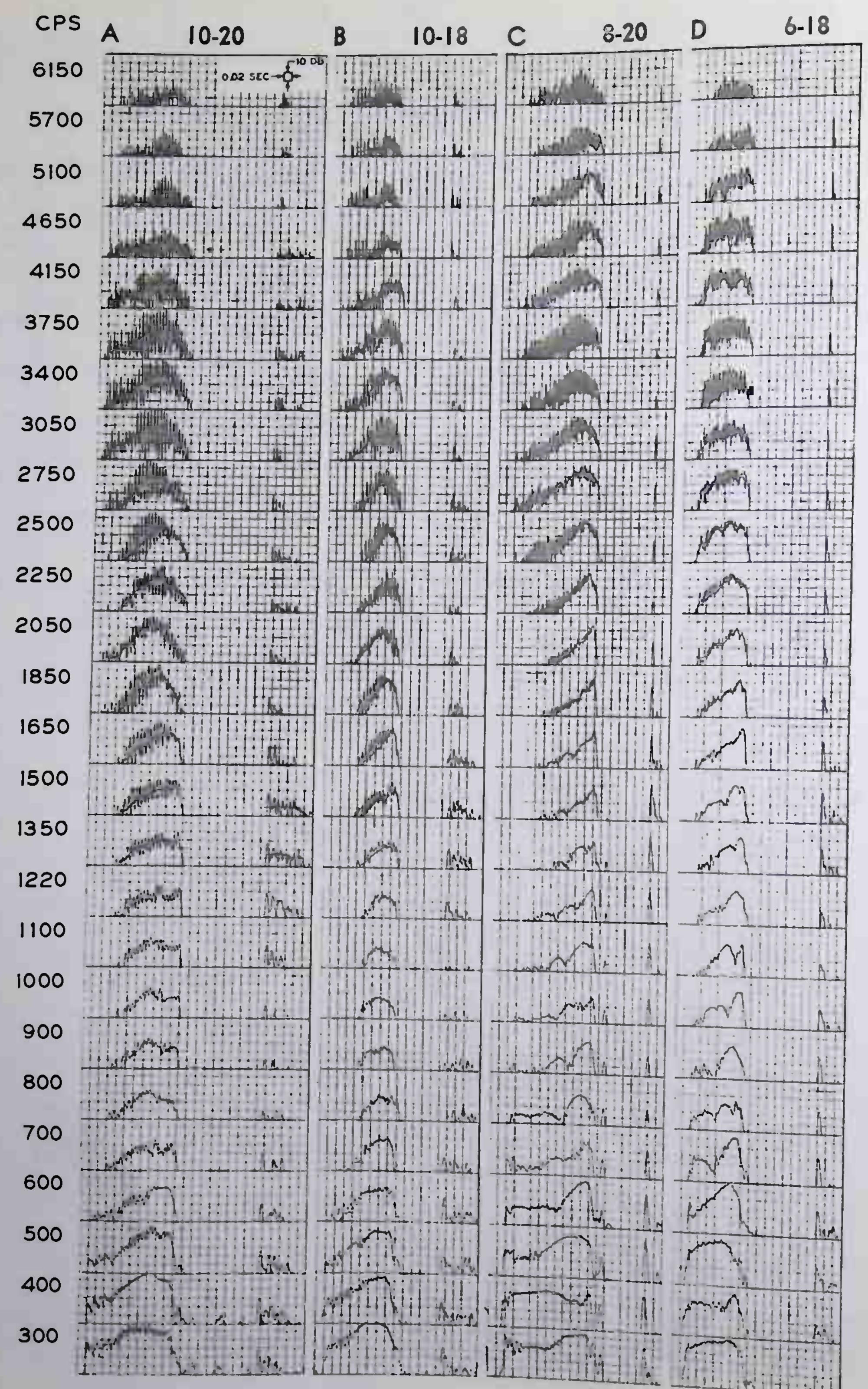
YIP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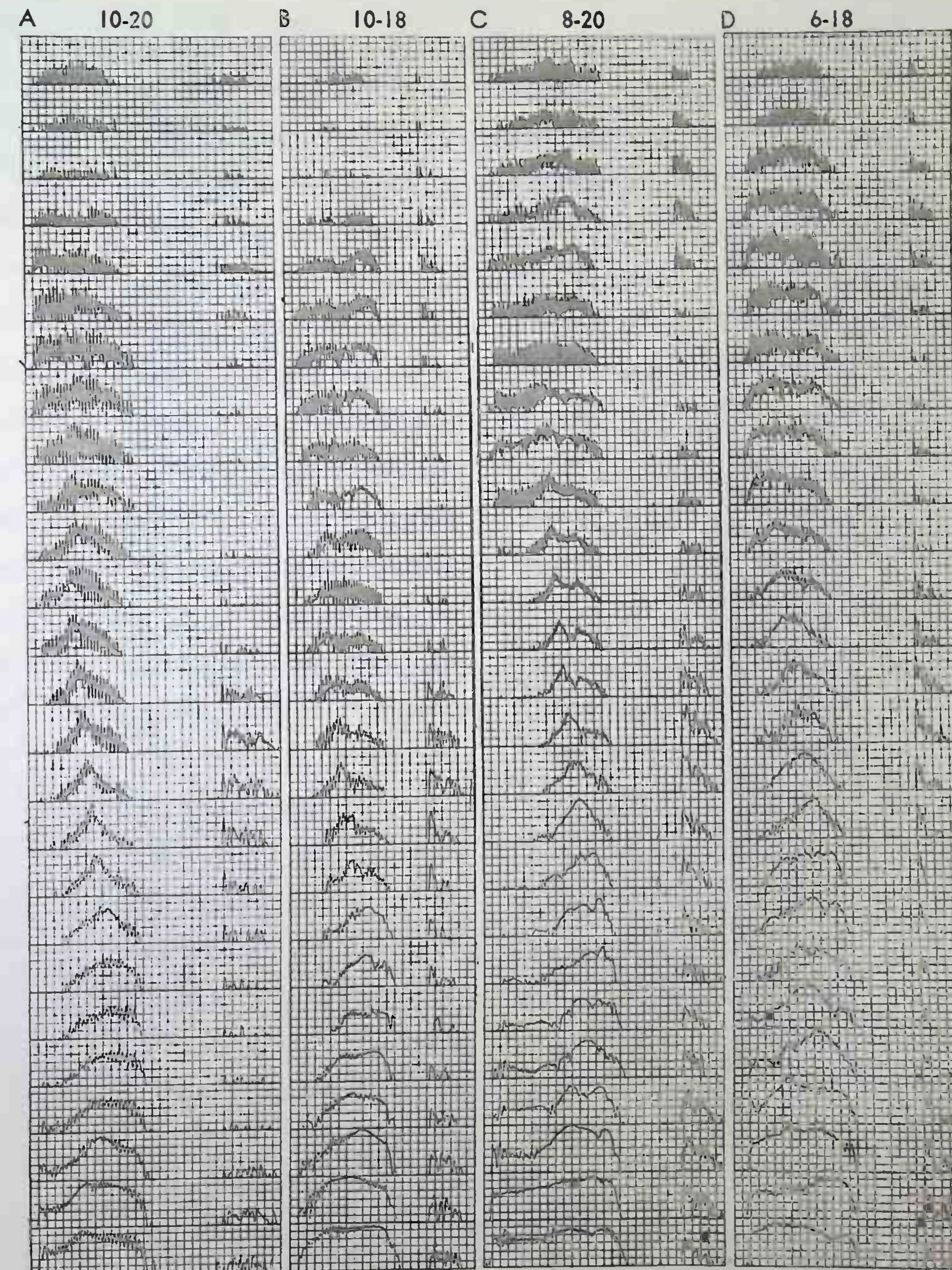
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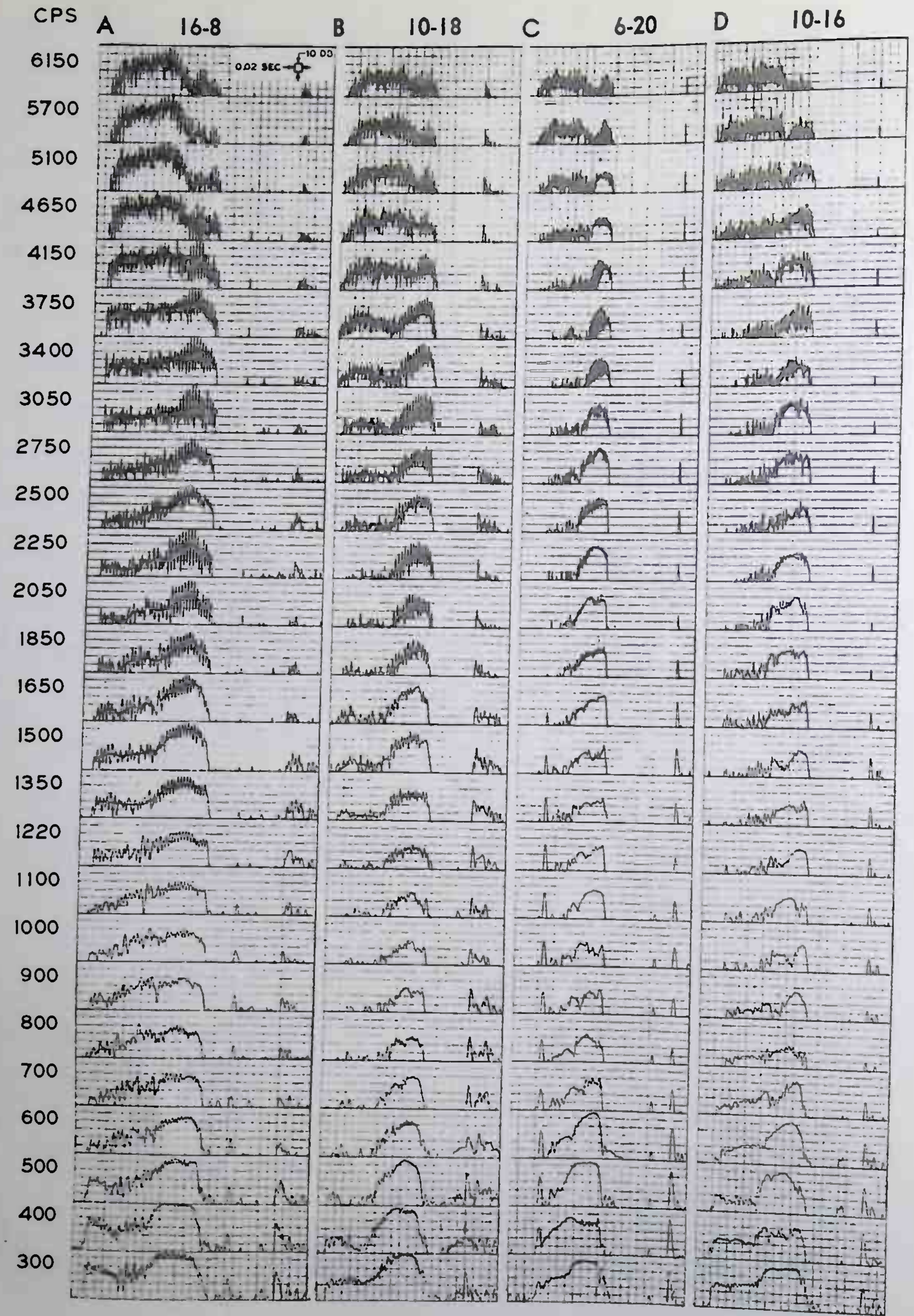
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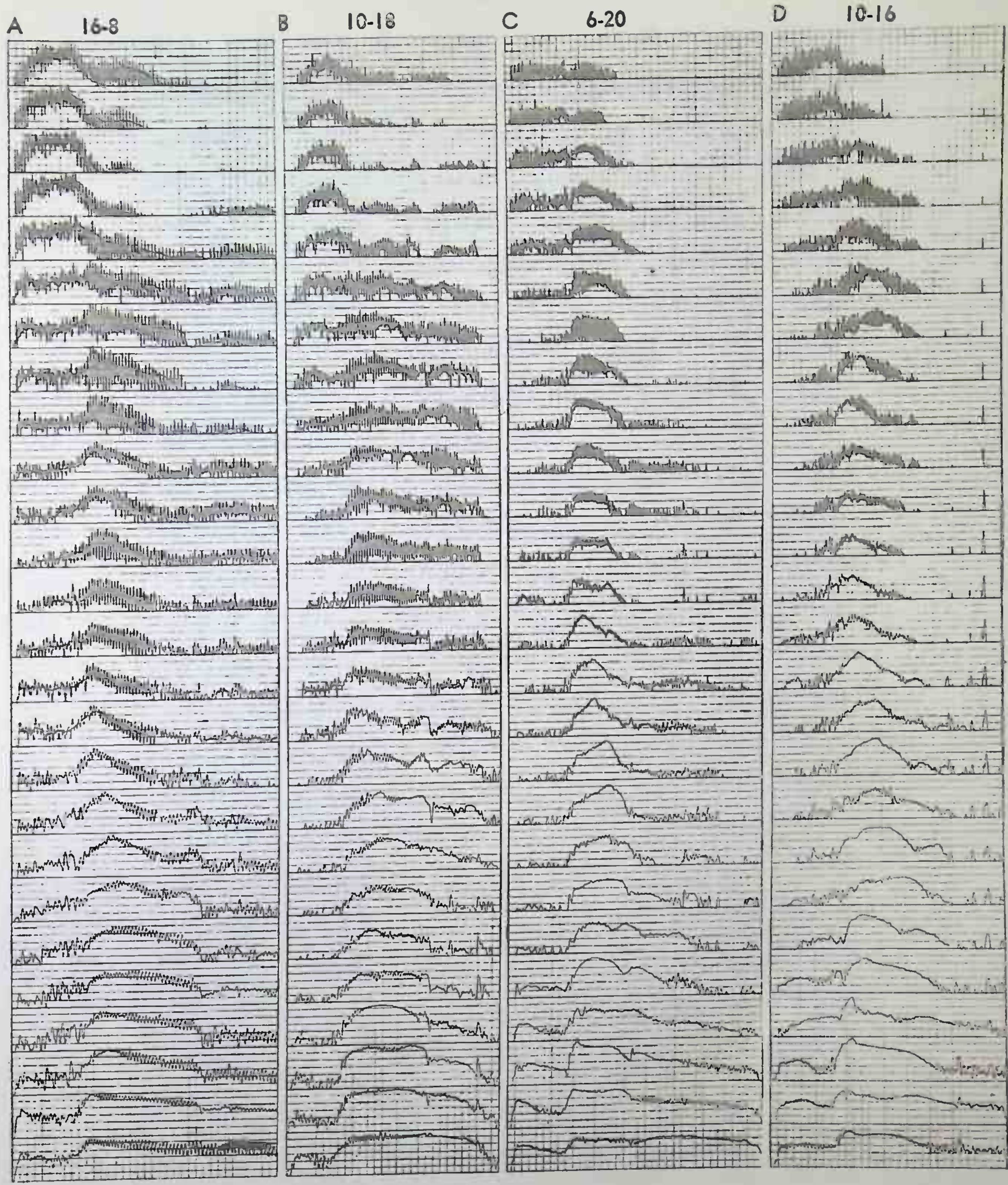
ZONE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C 6-20
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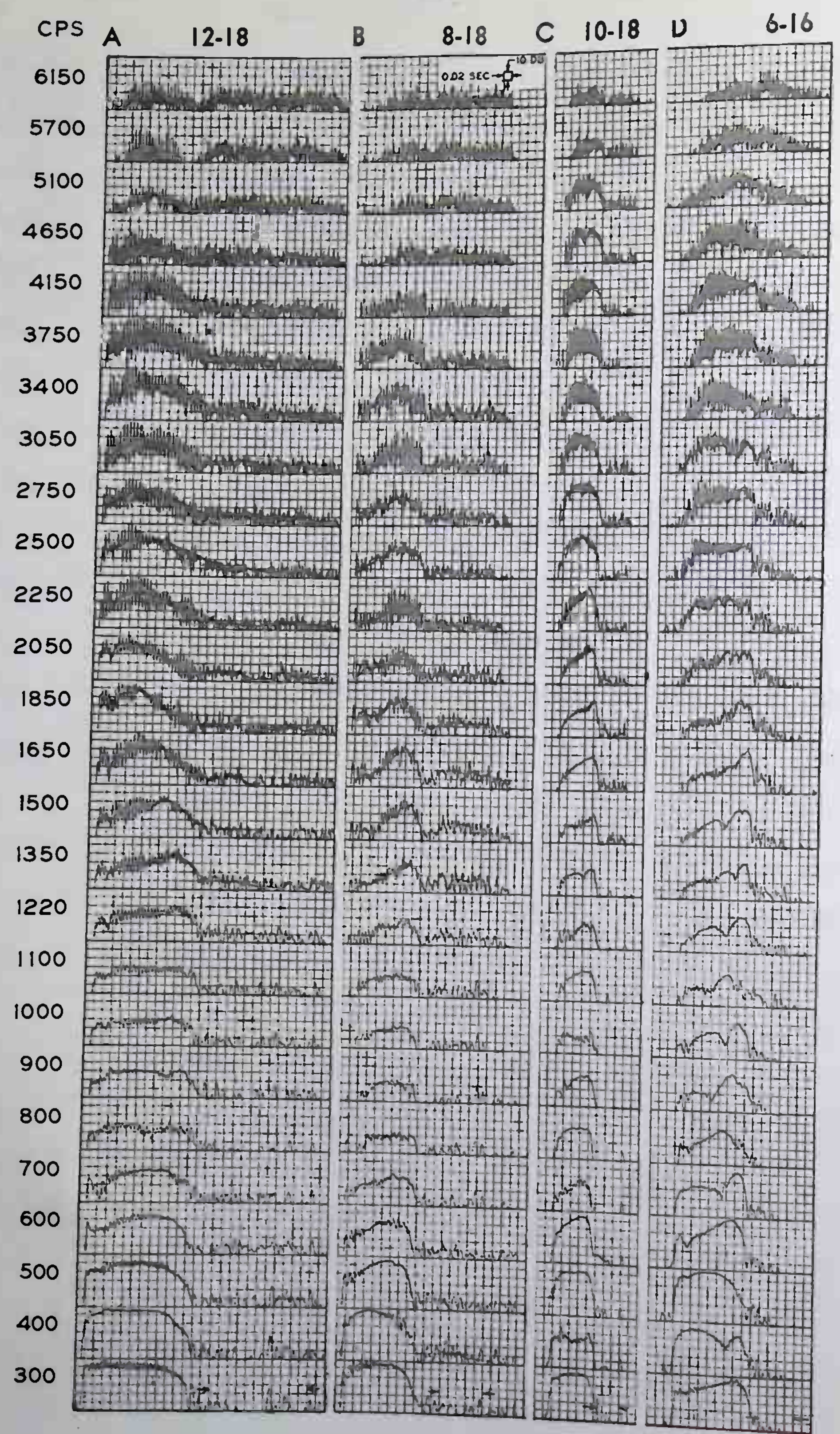
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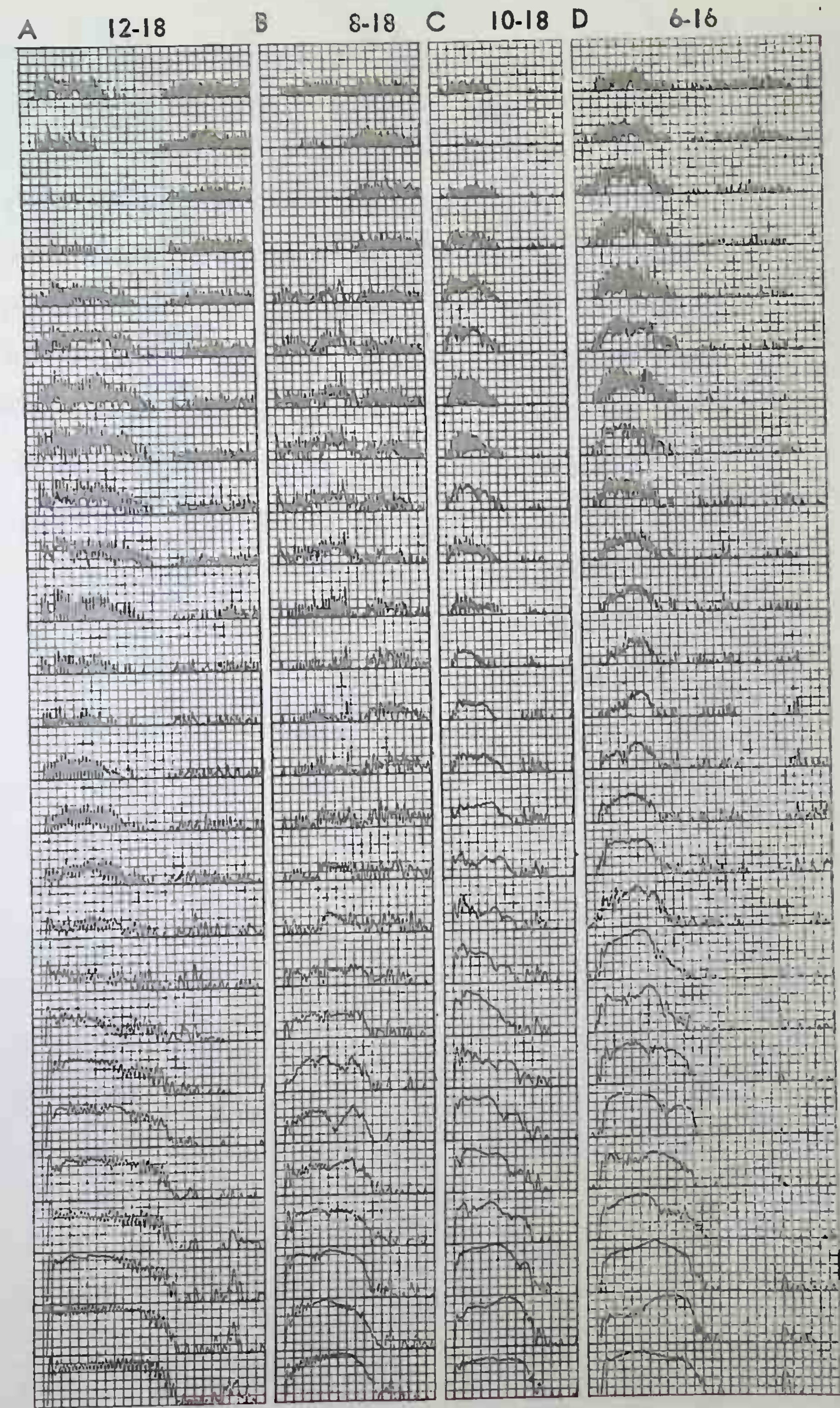
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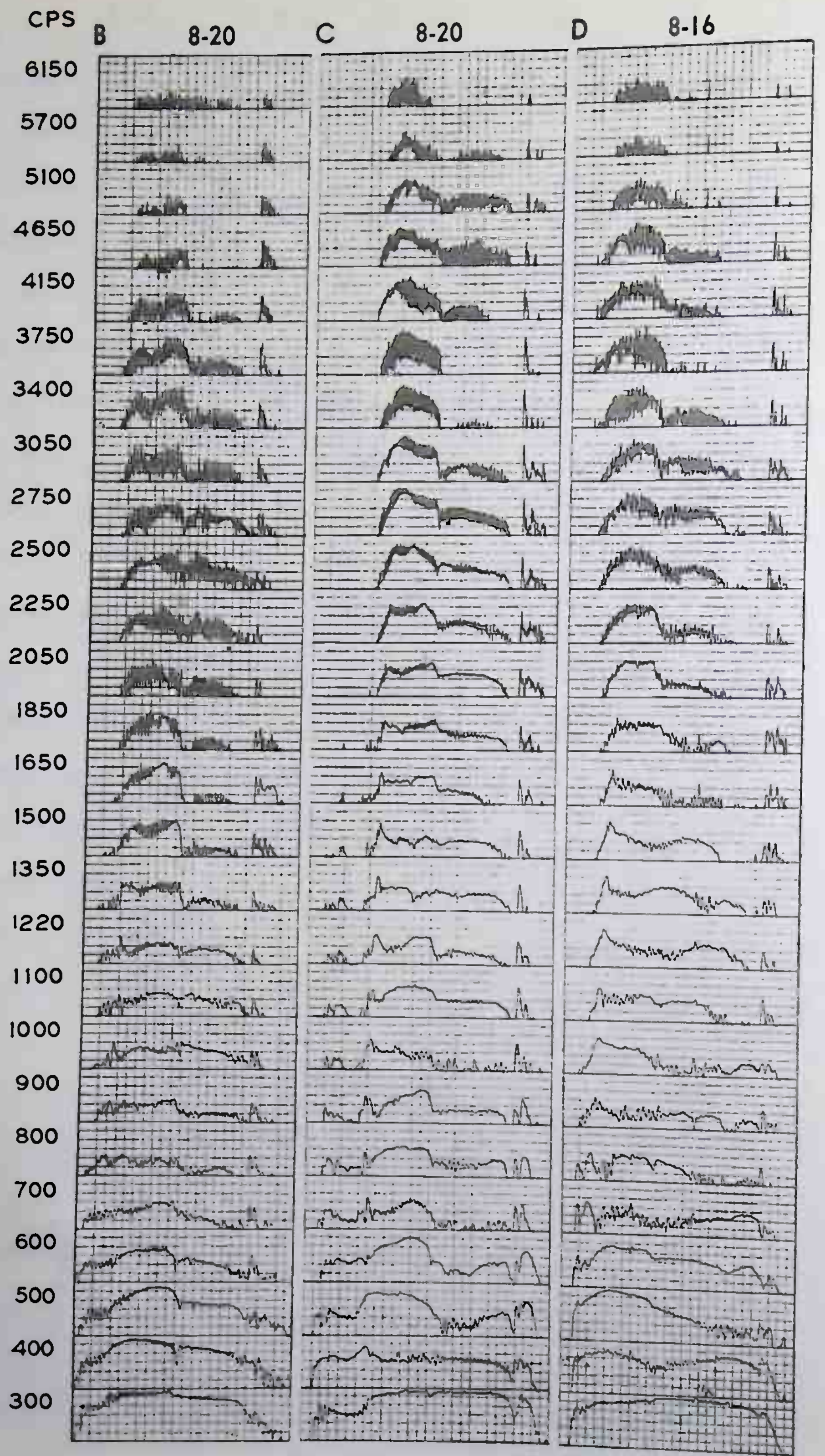
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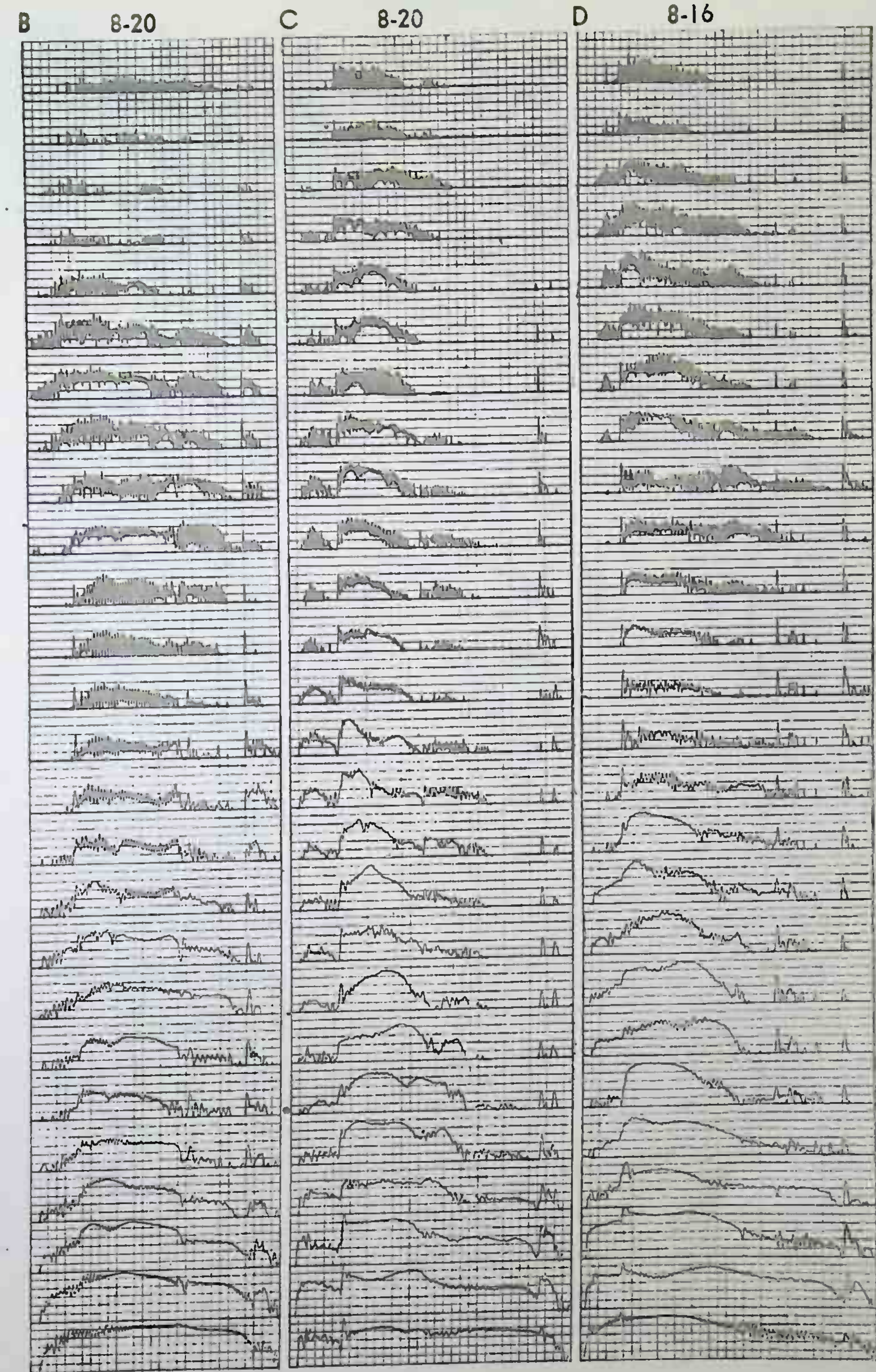
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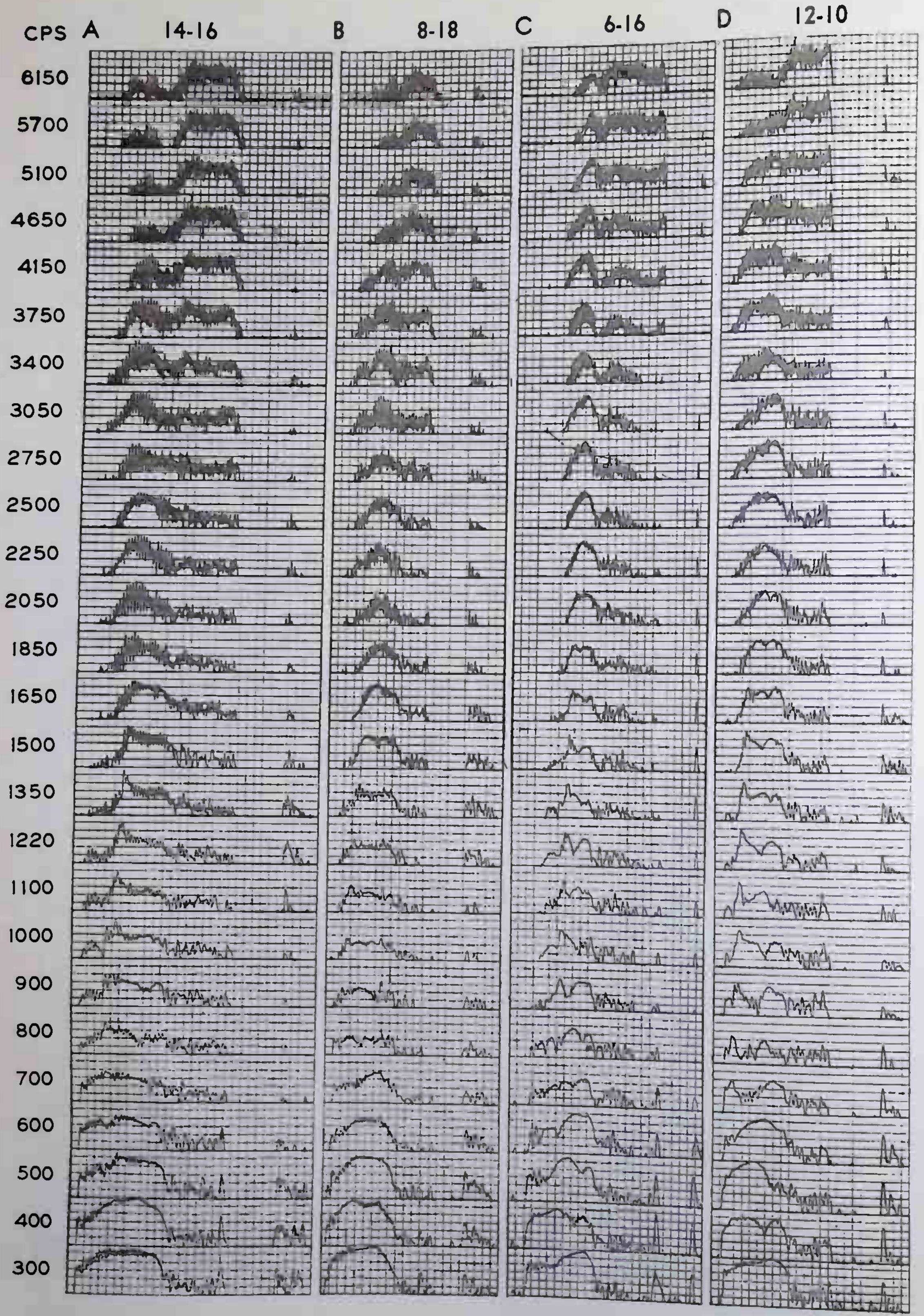
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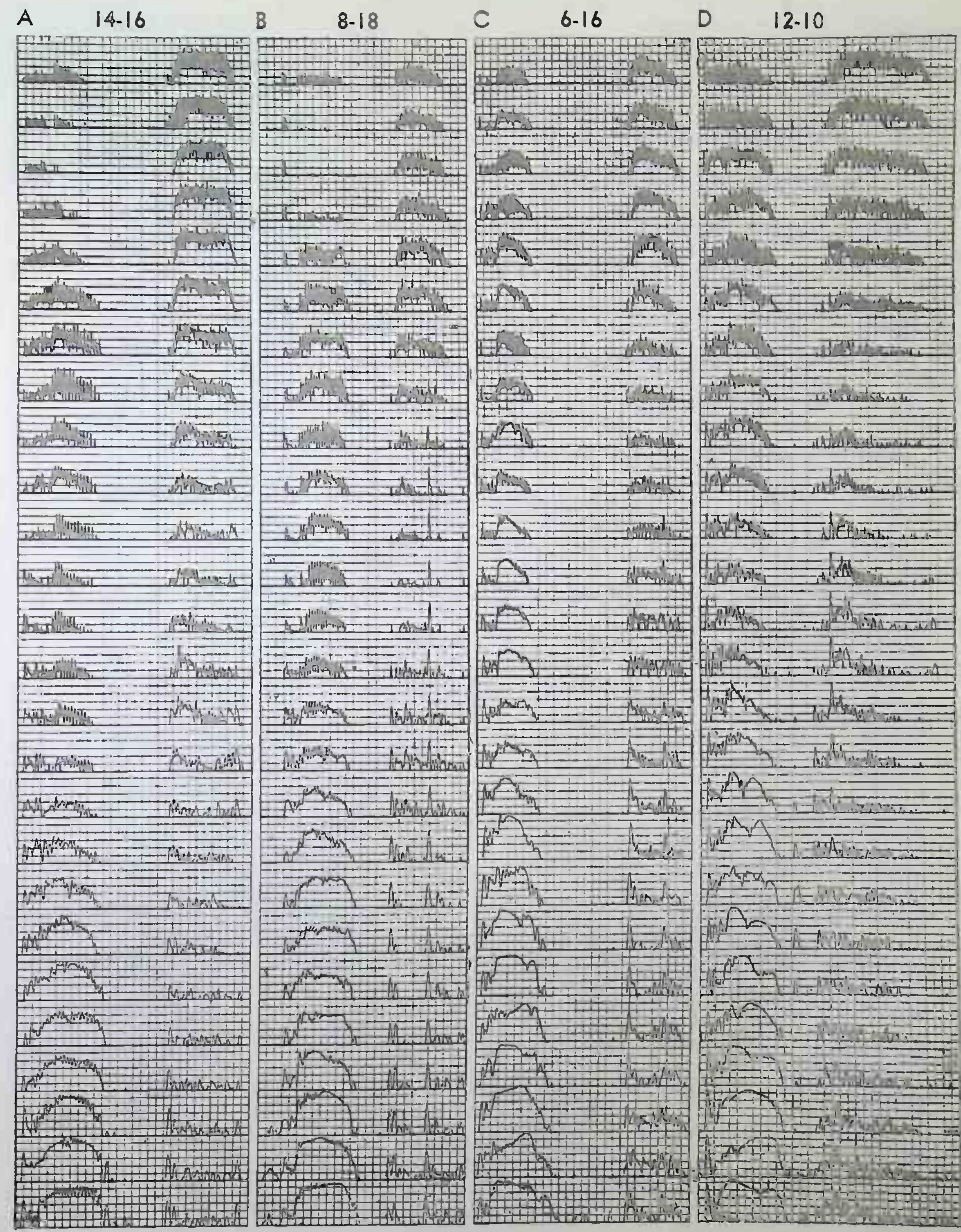
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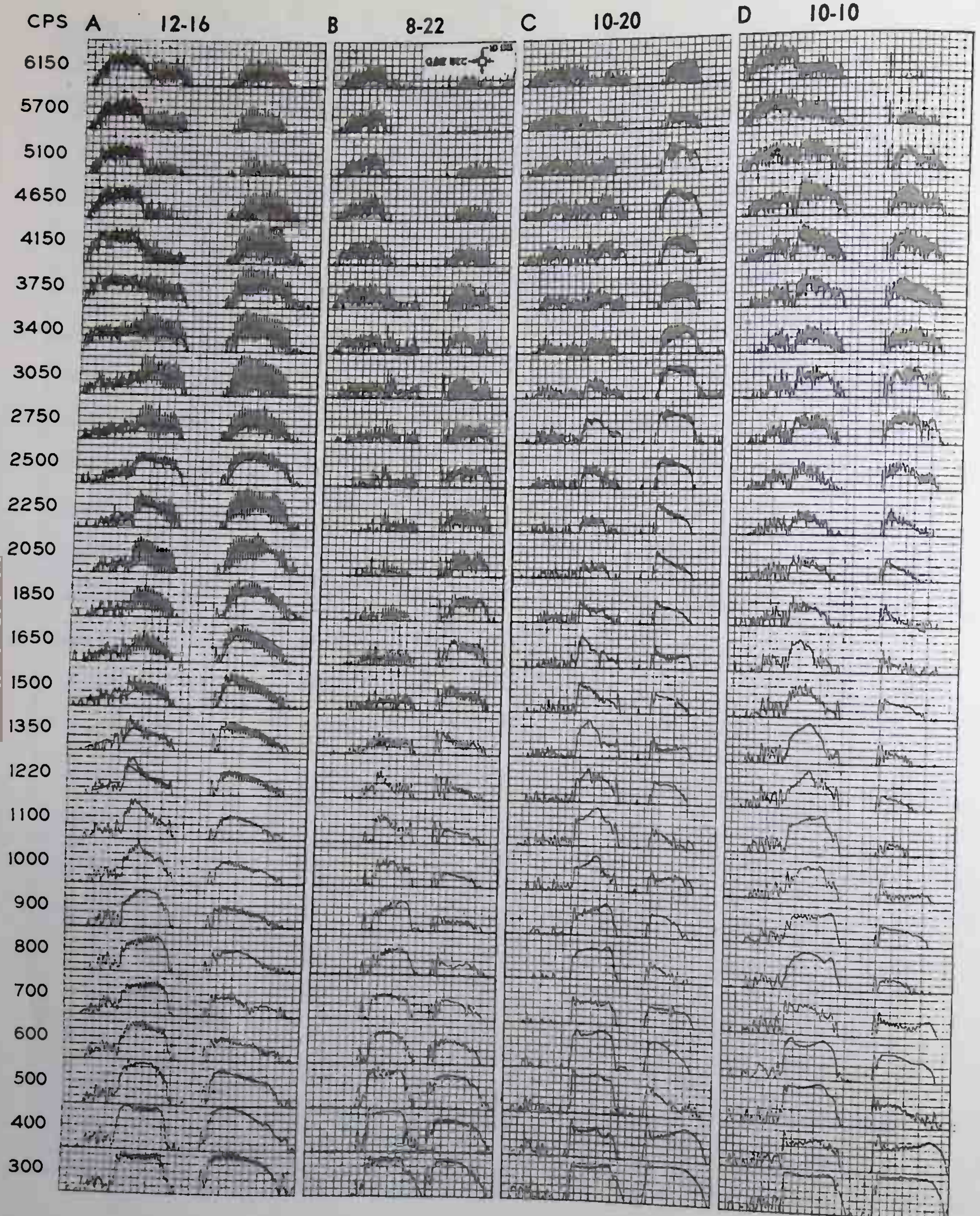

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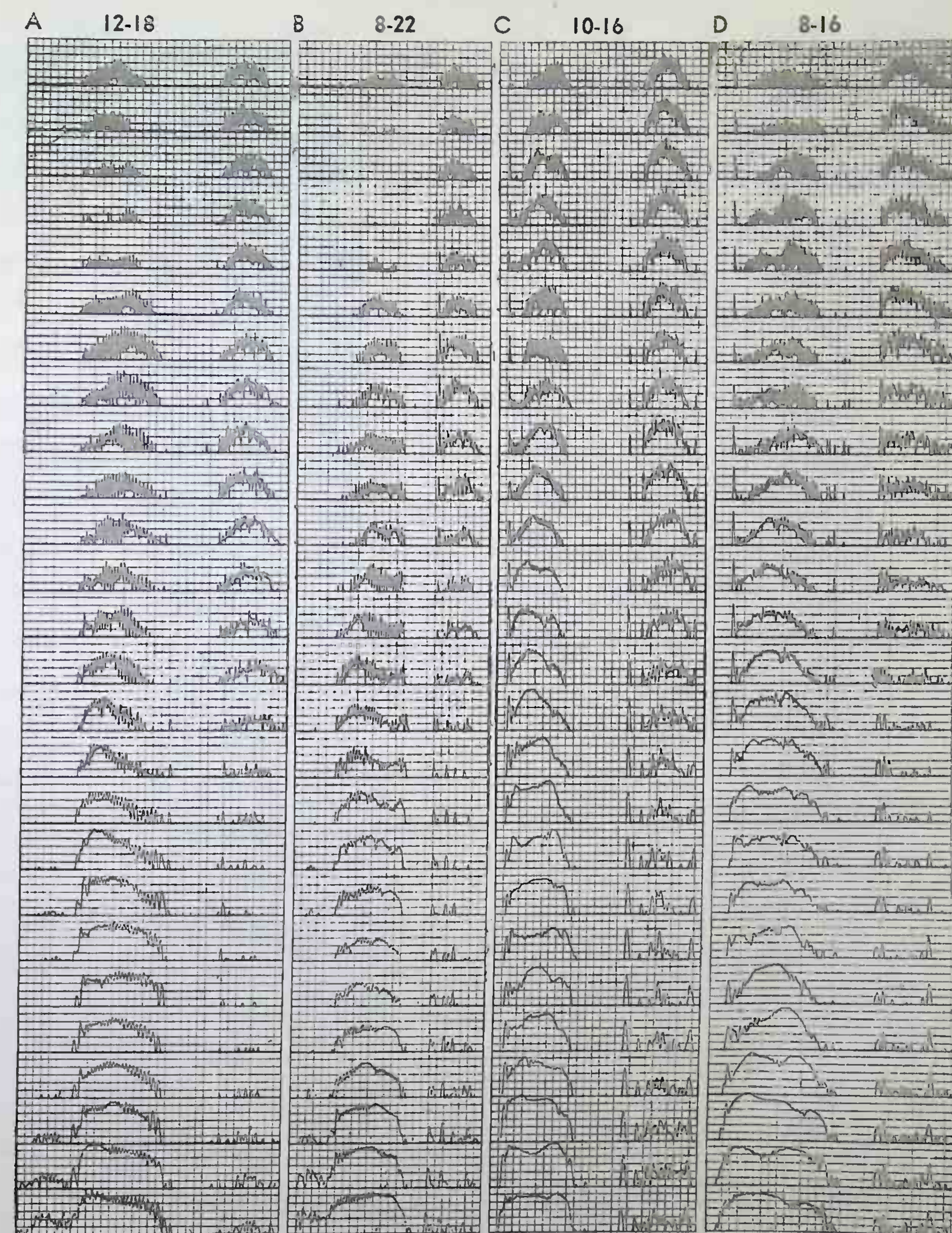

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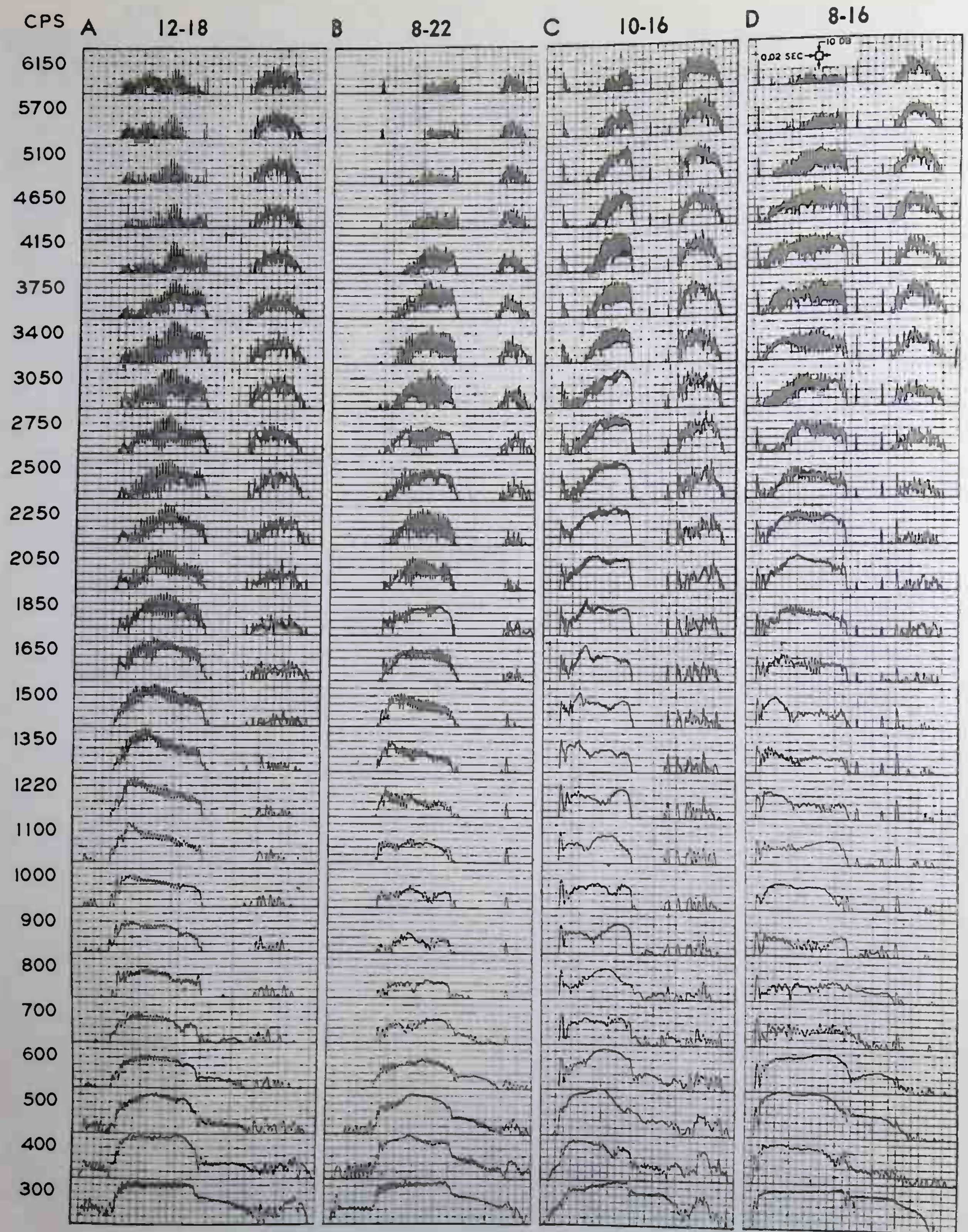

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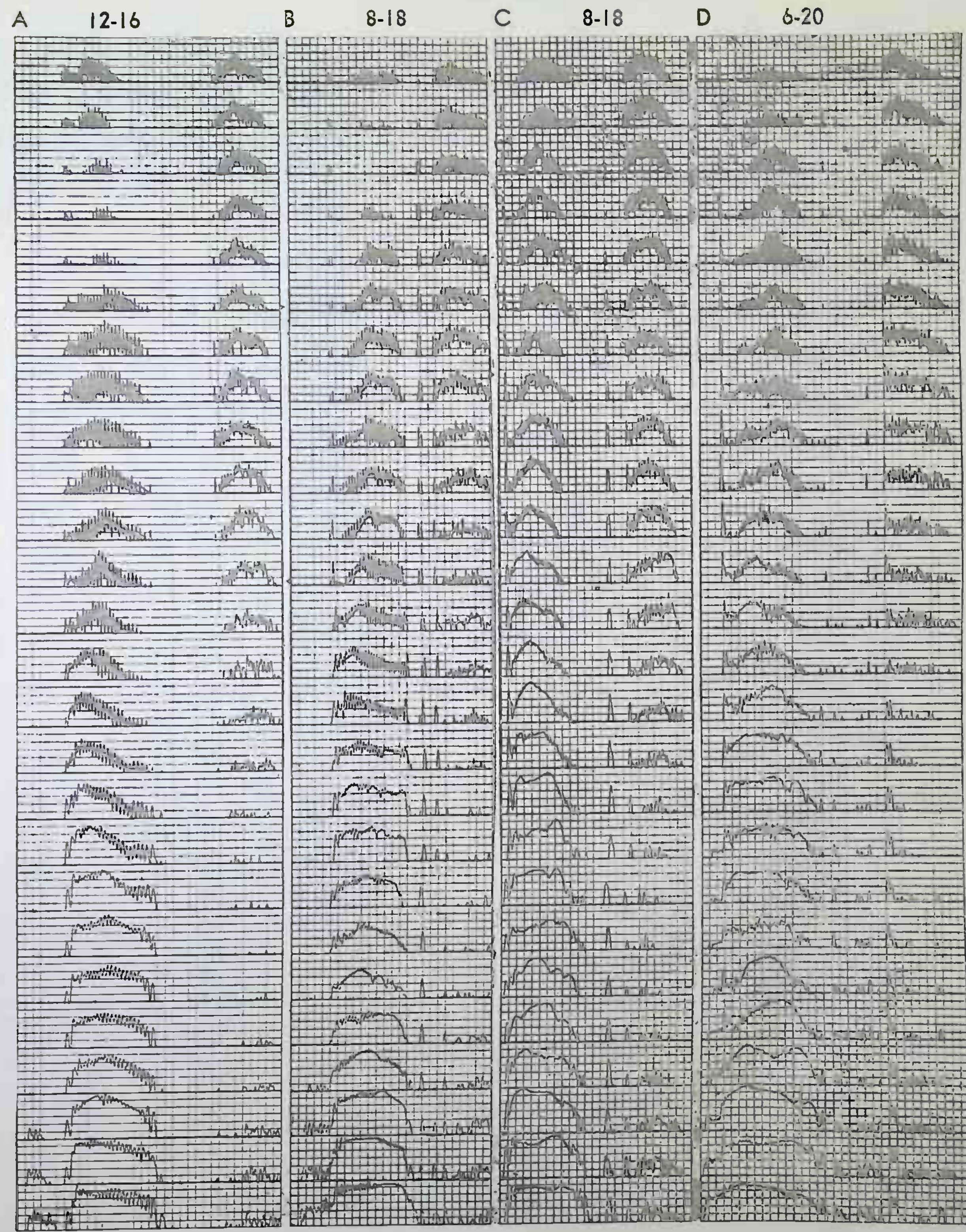

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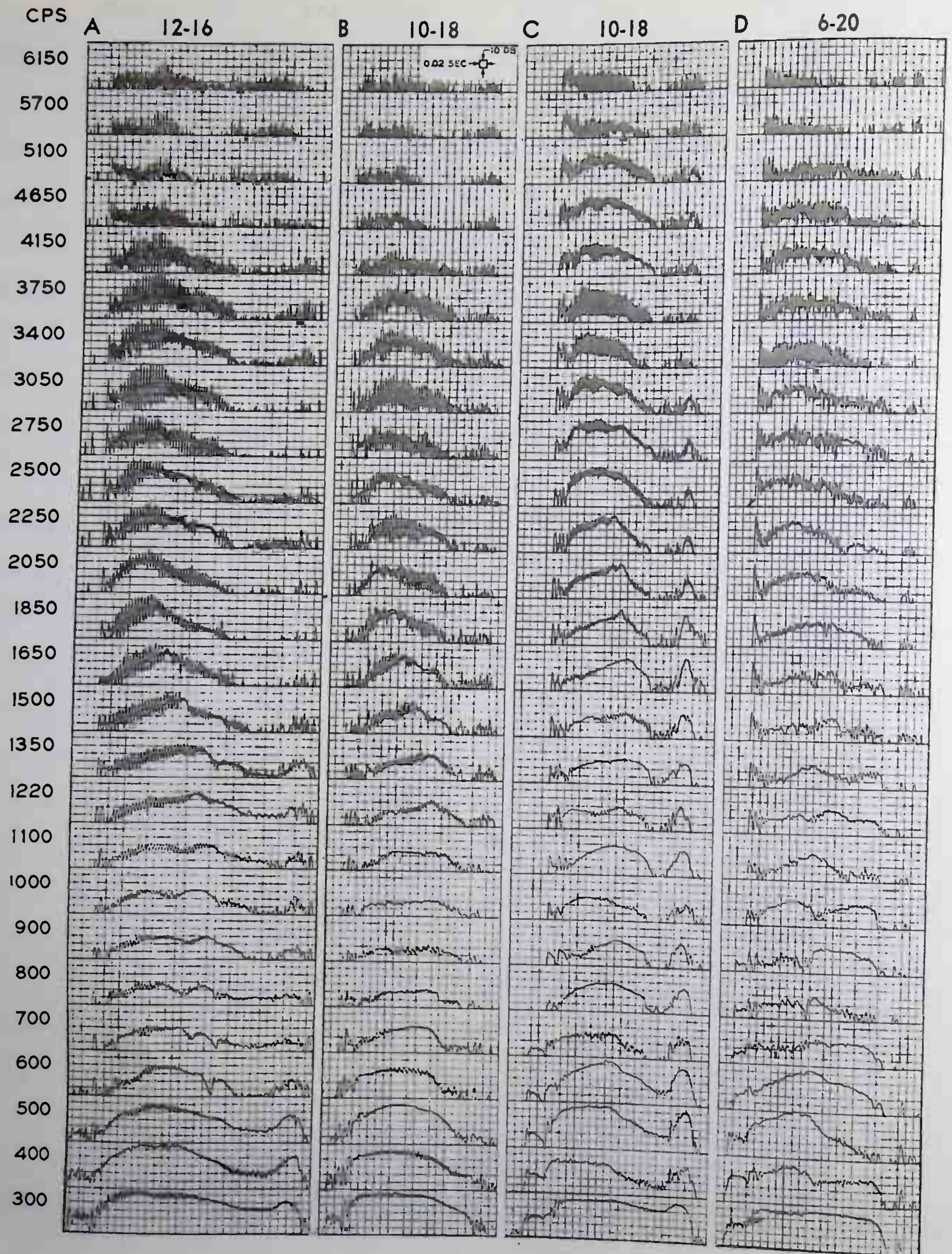
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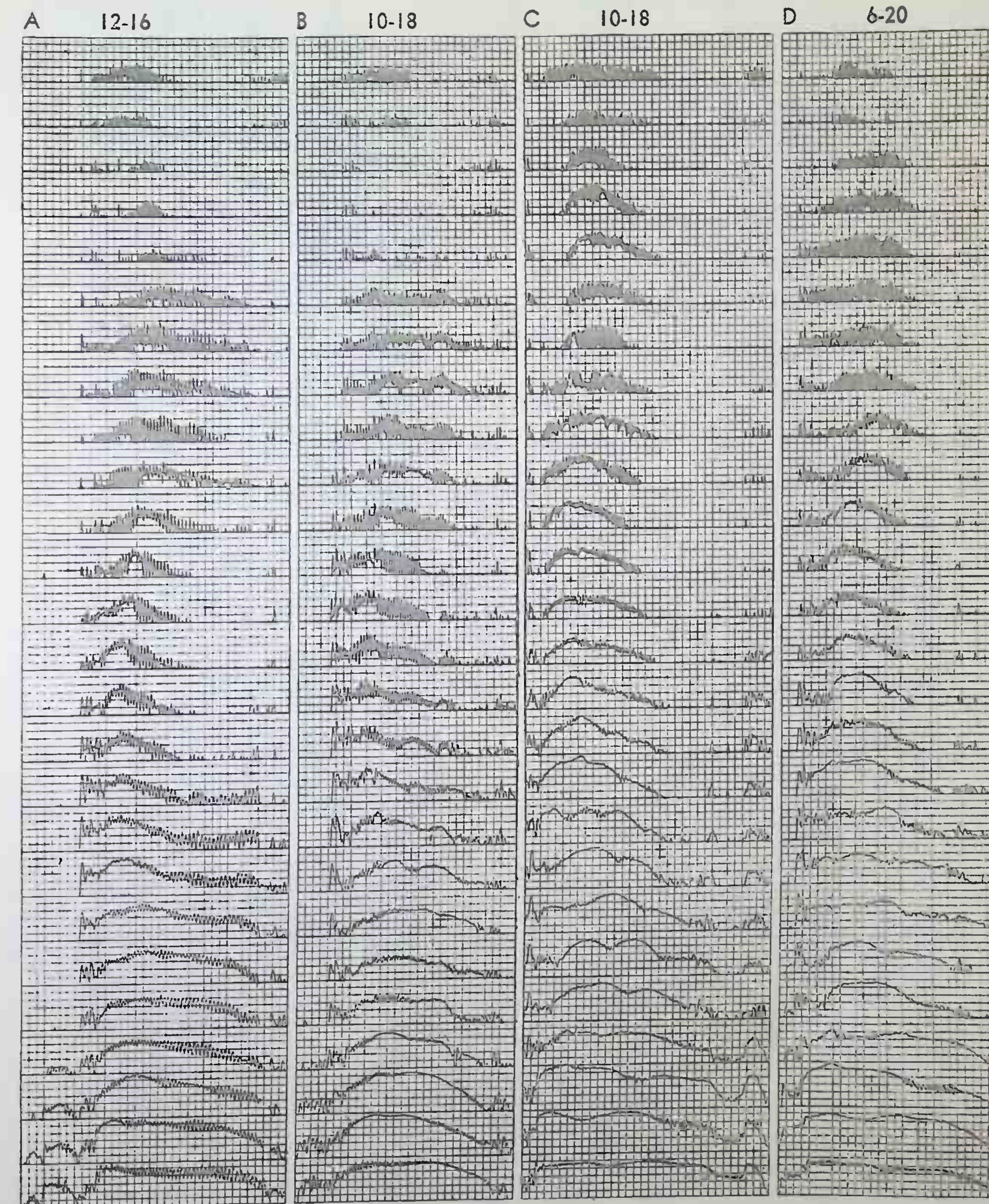
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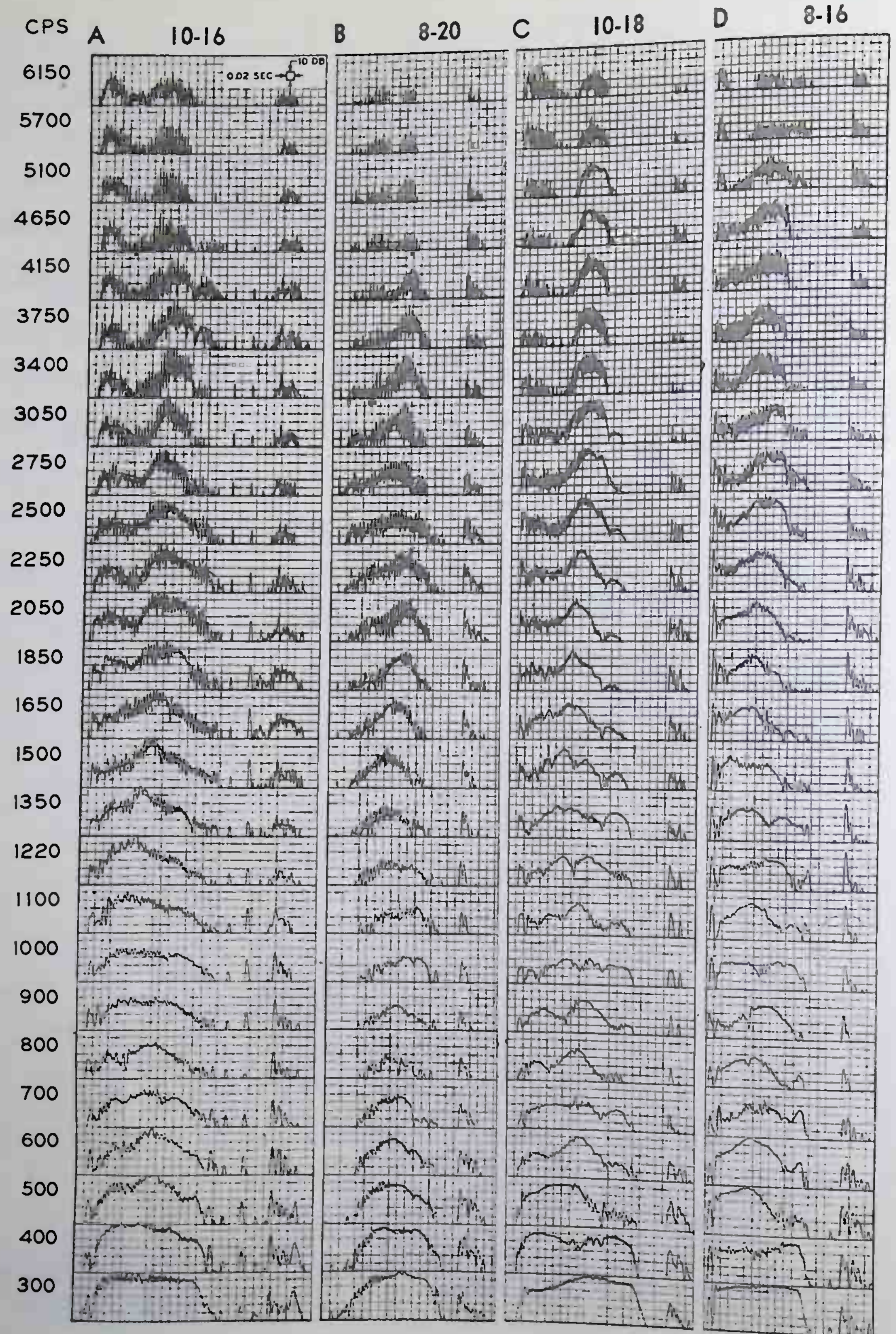
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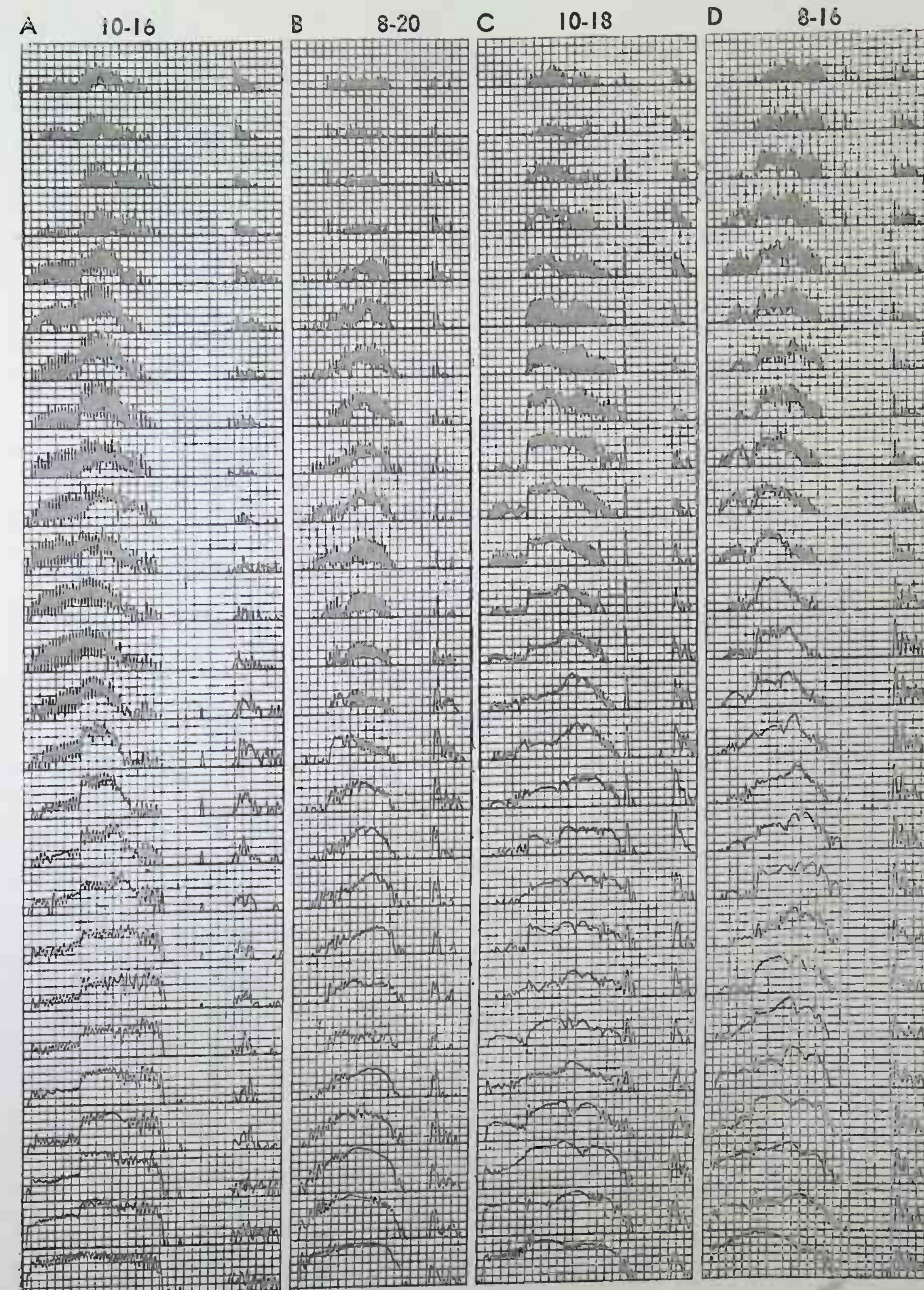
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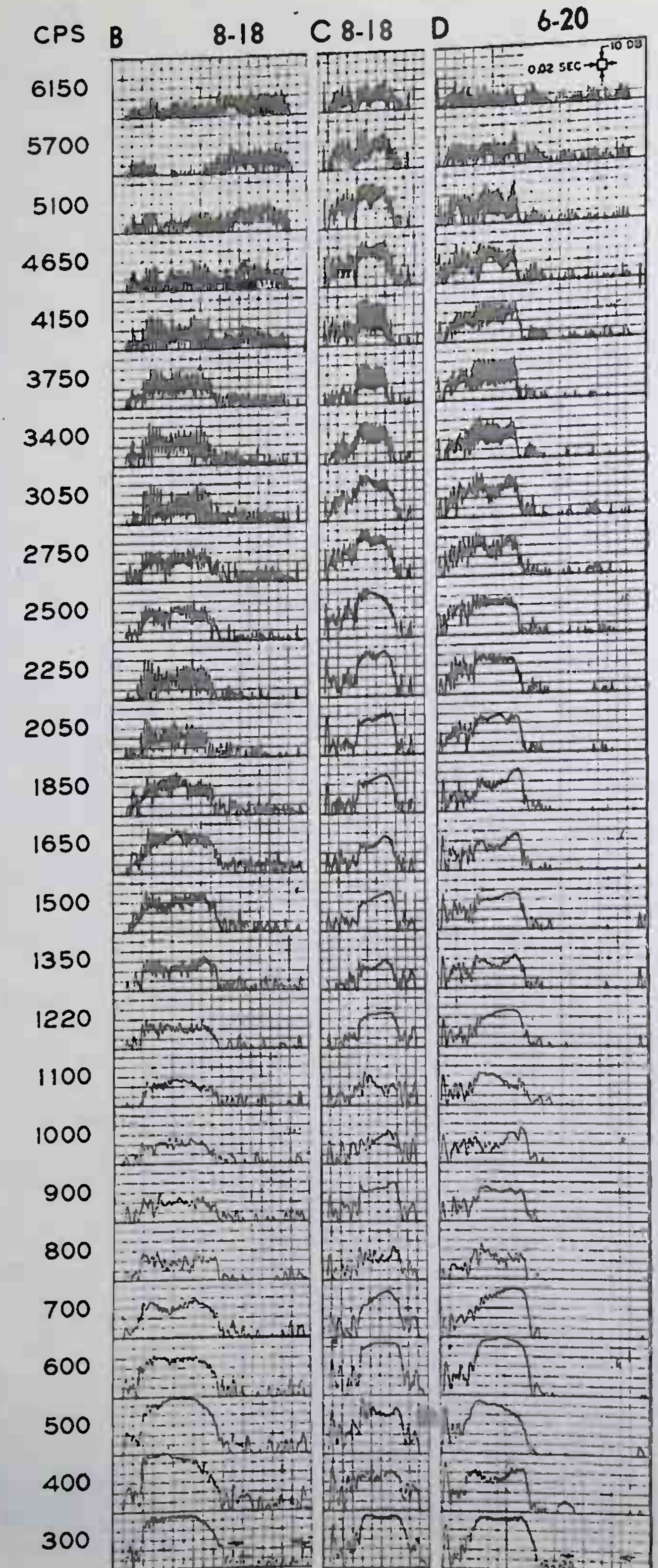
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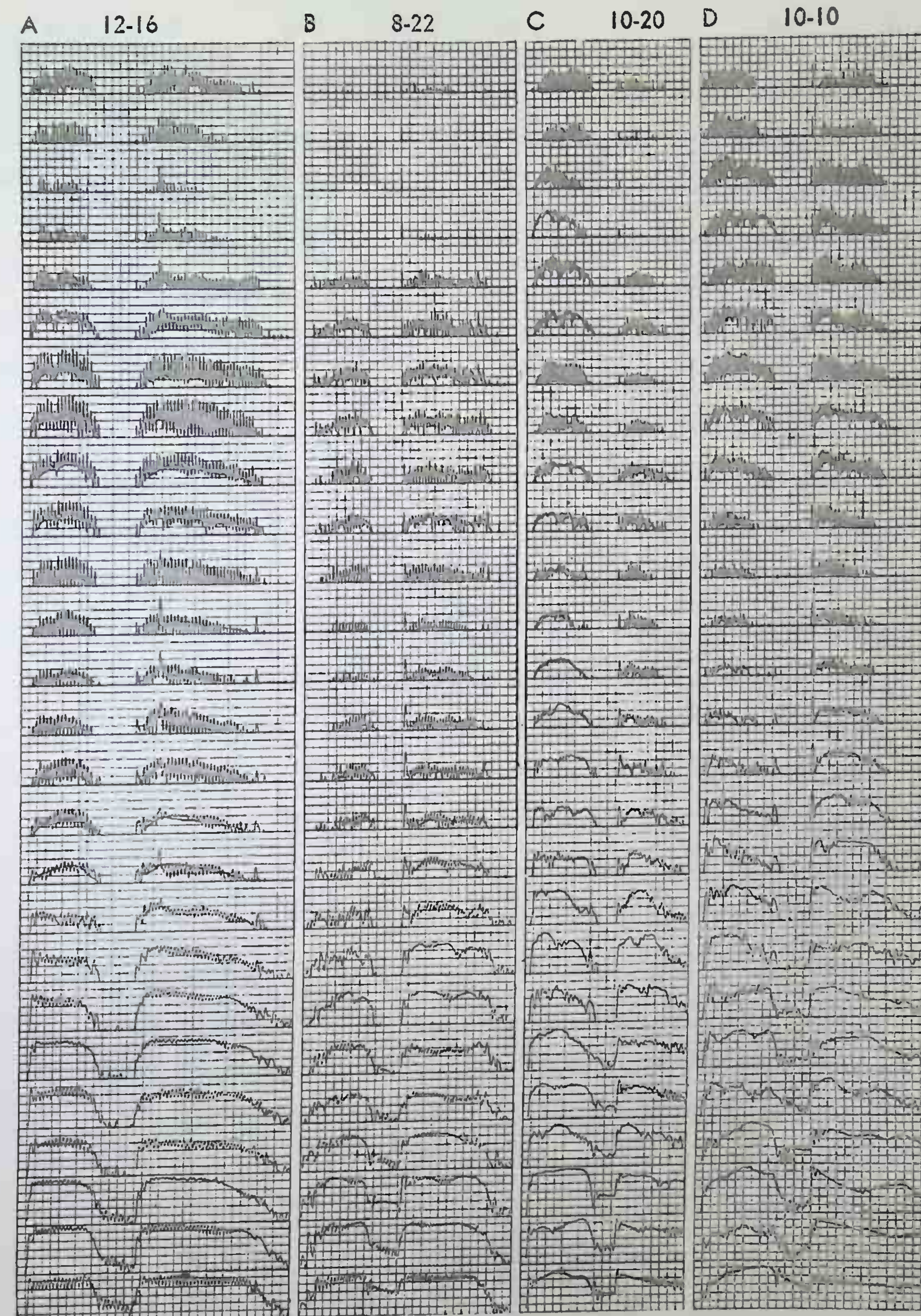
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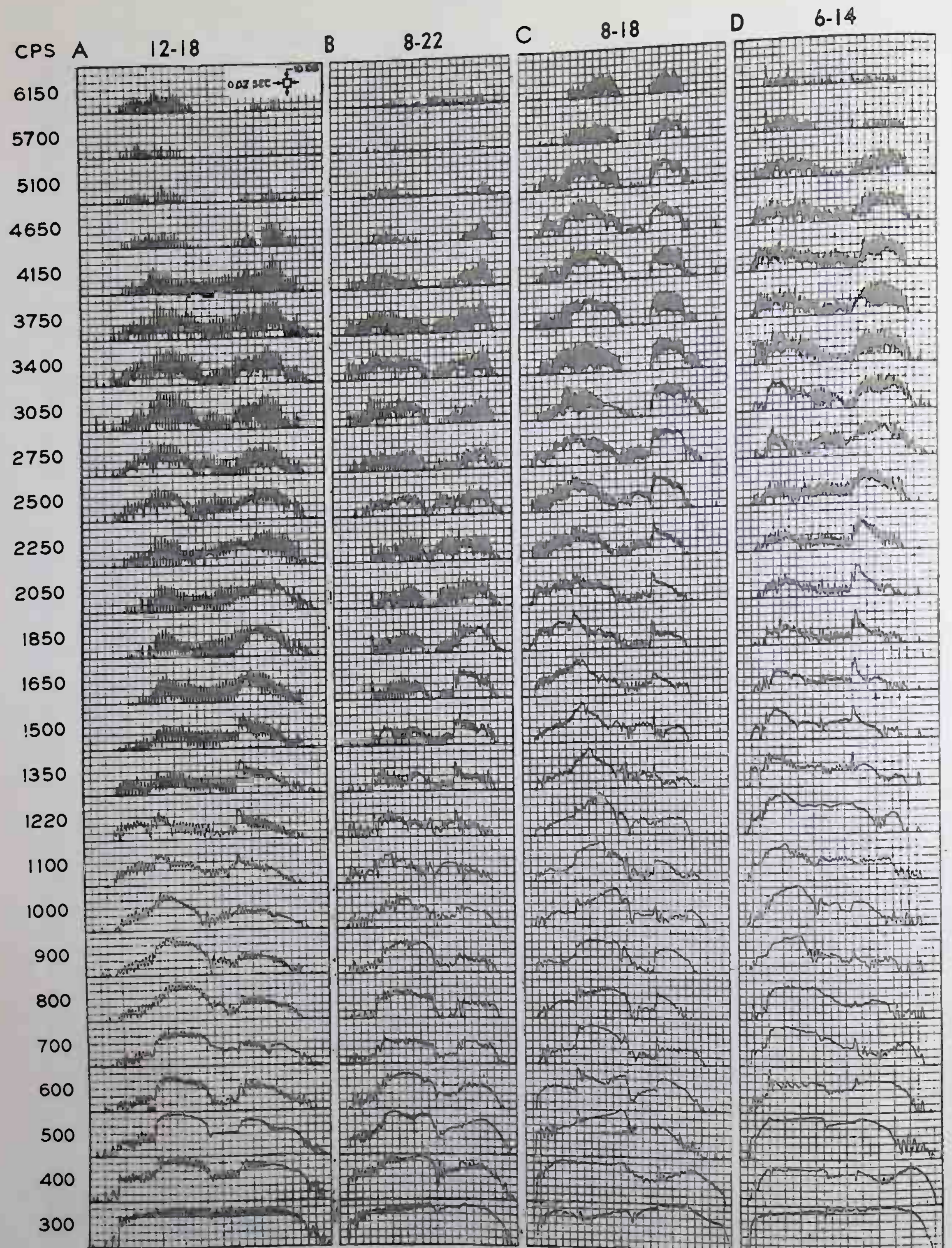
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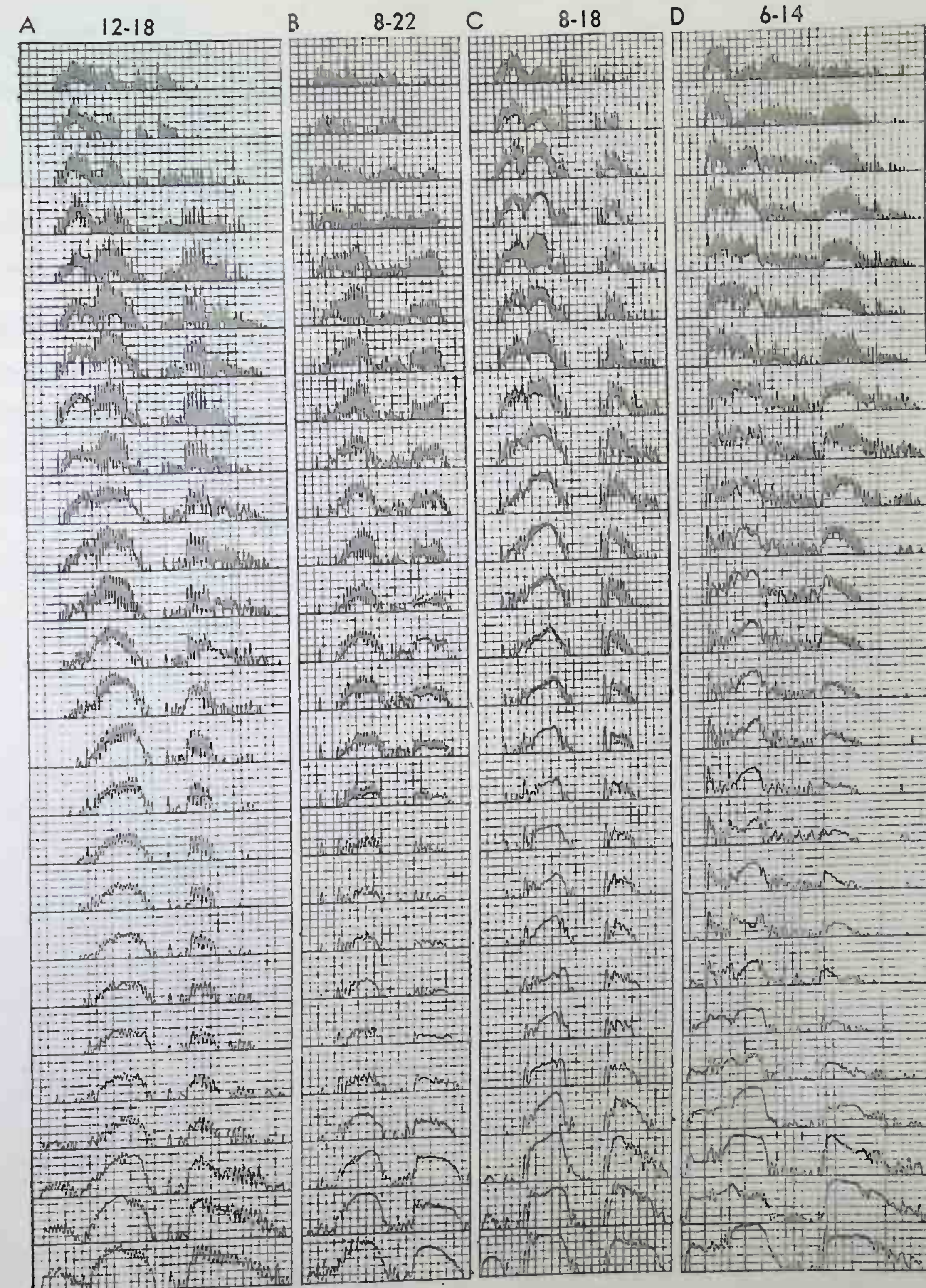
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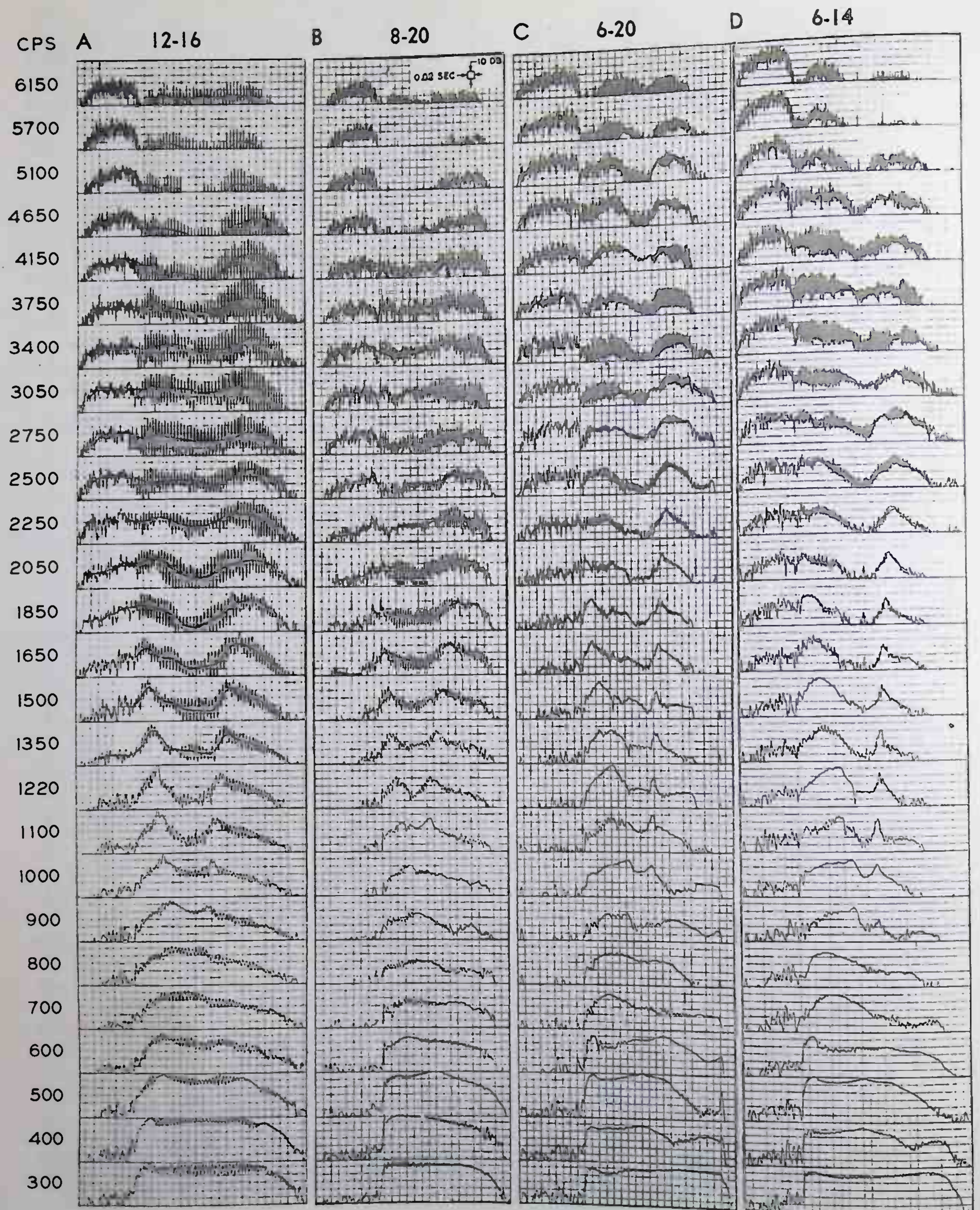

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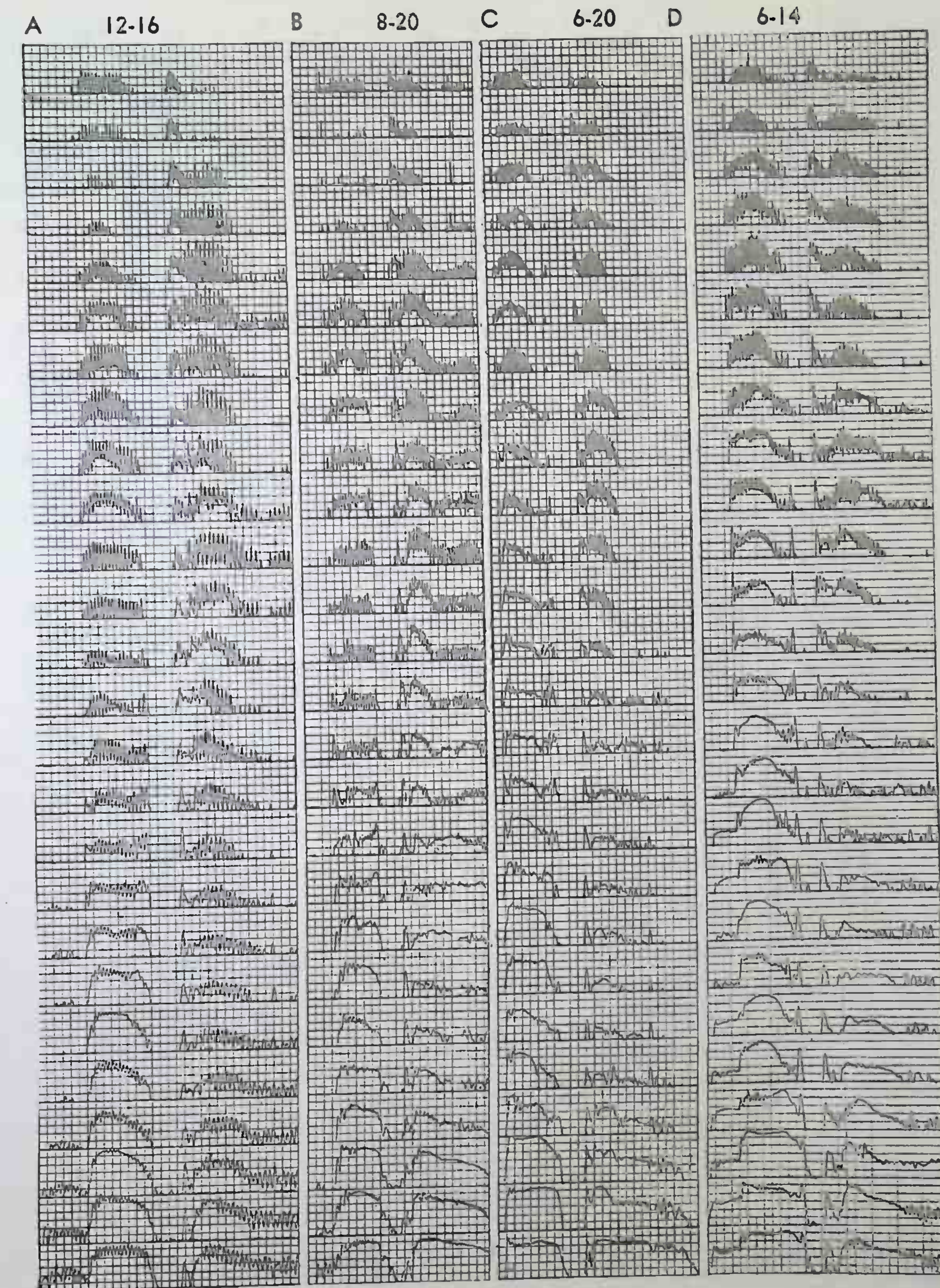

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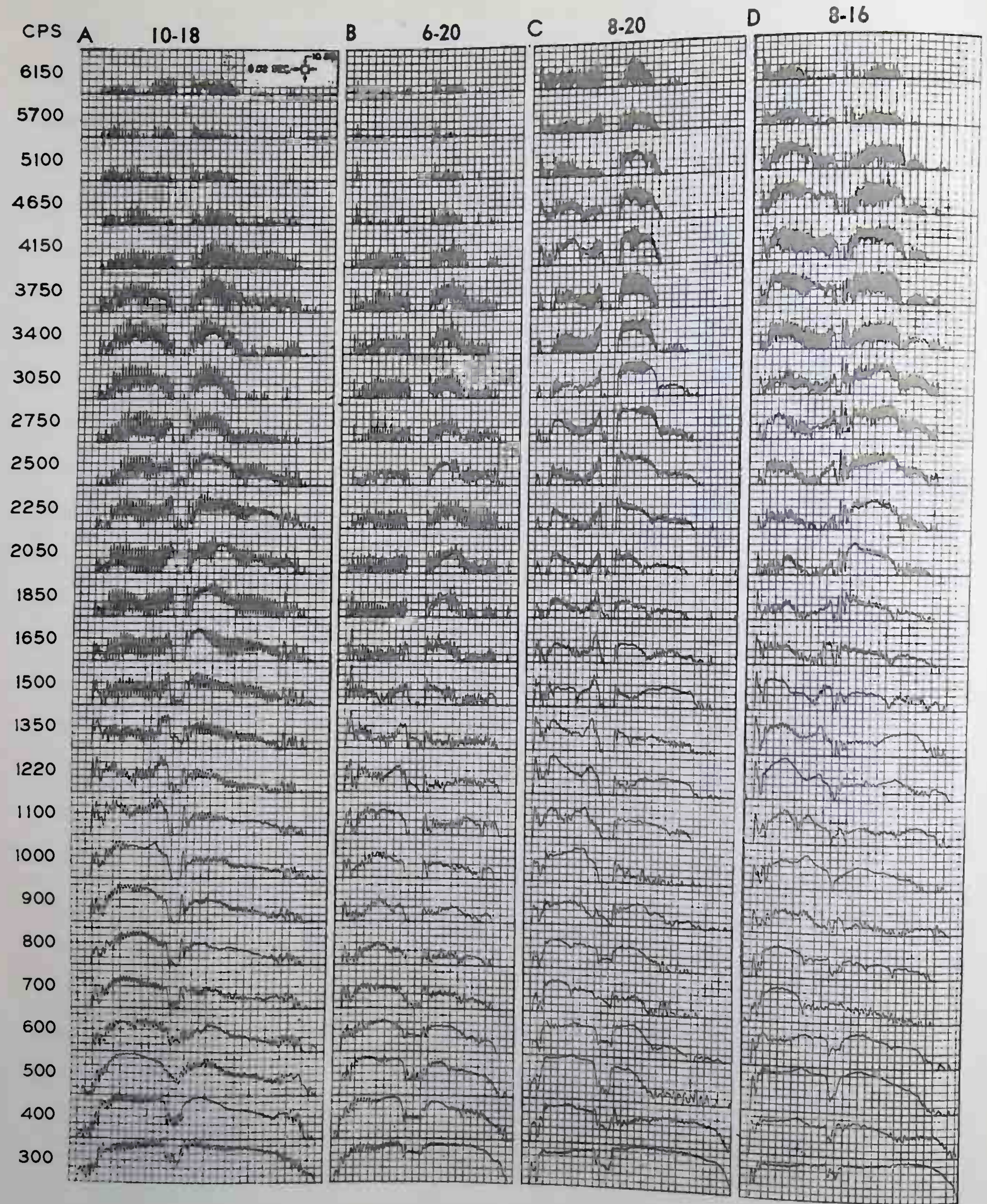

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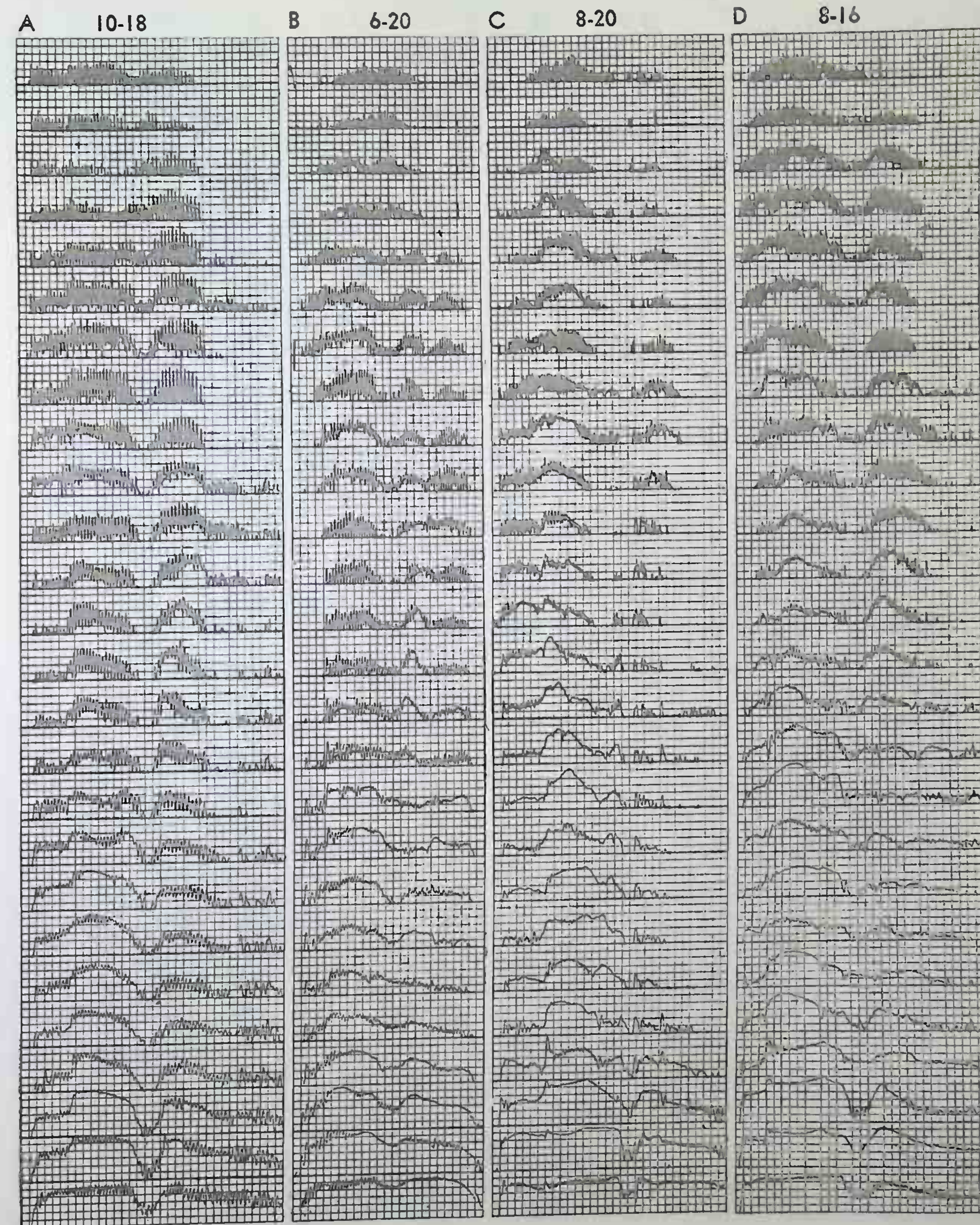

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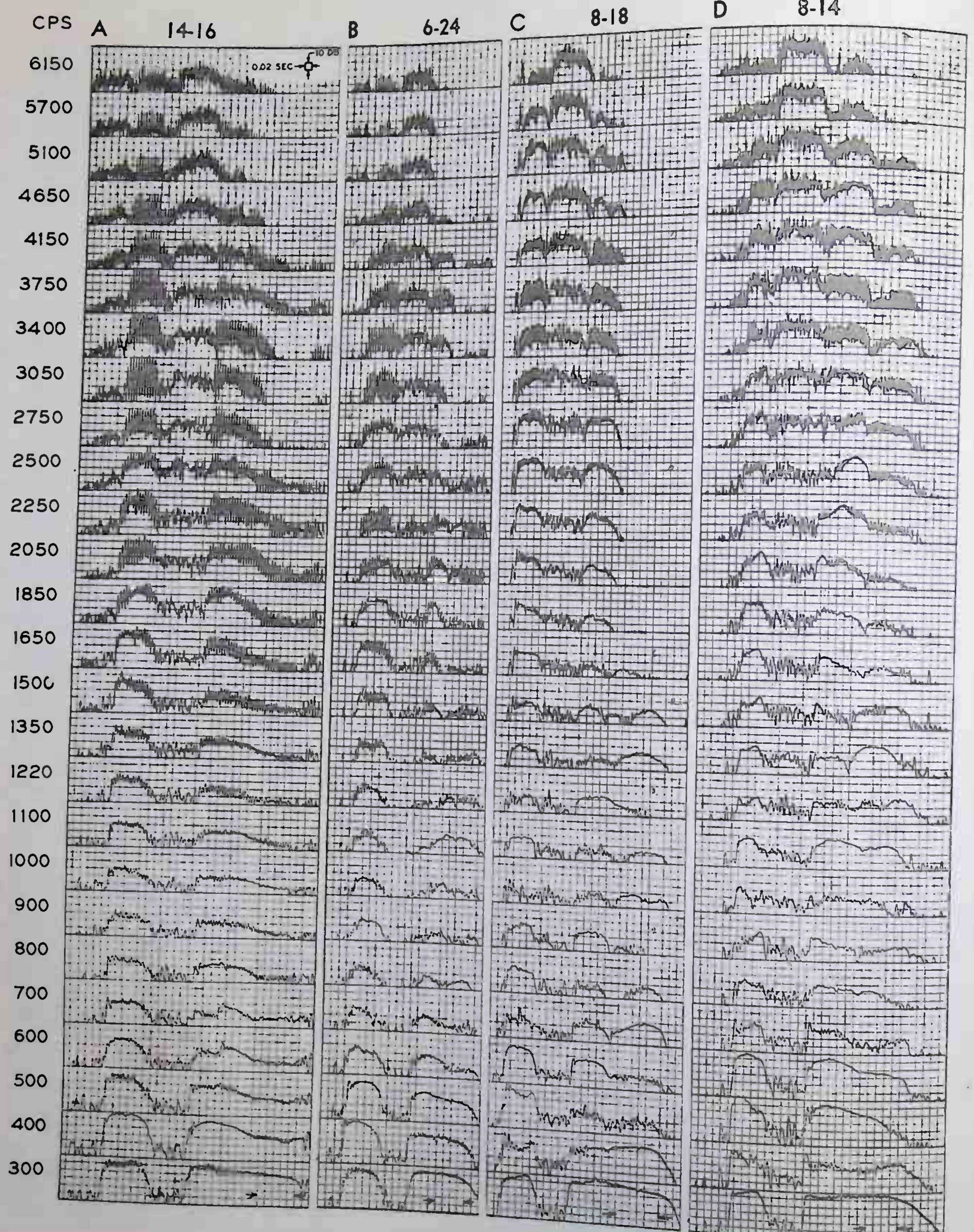

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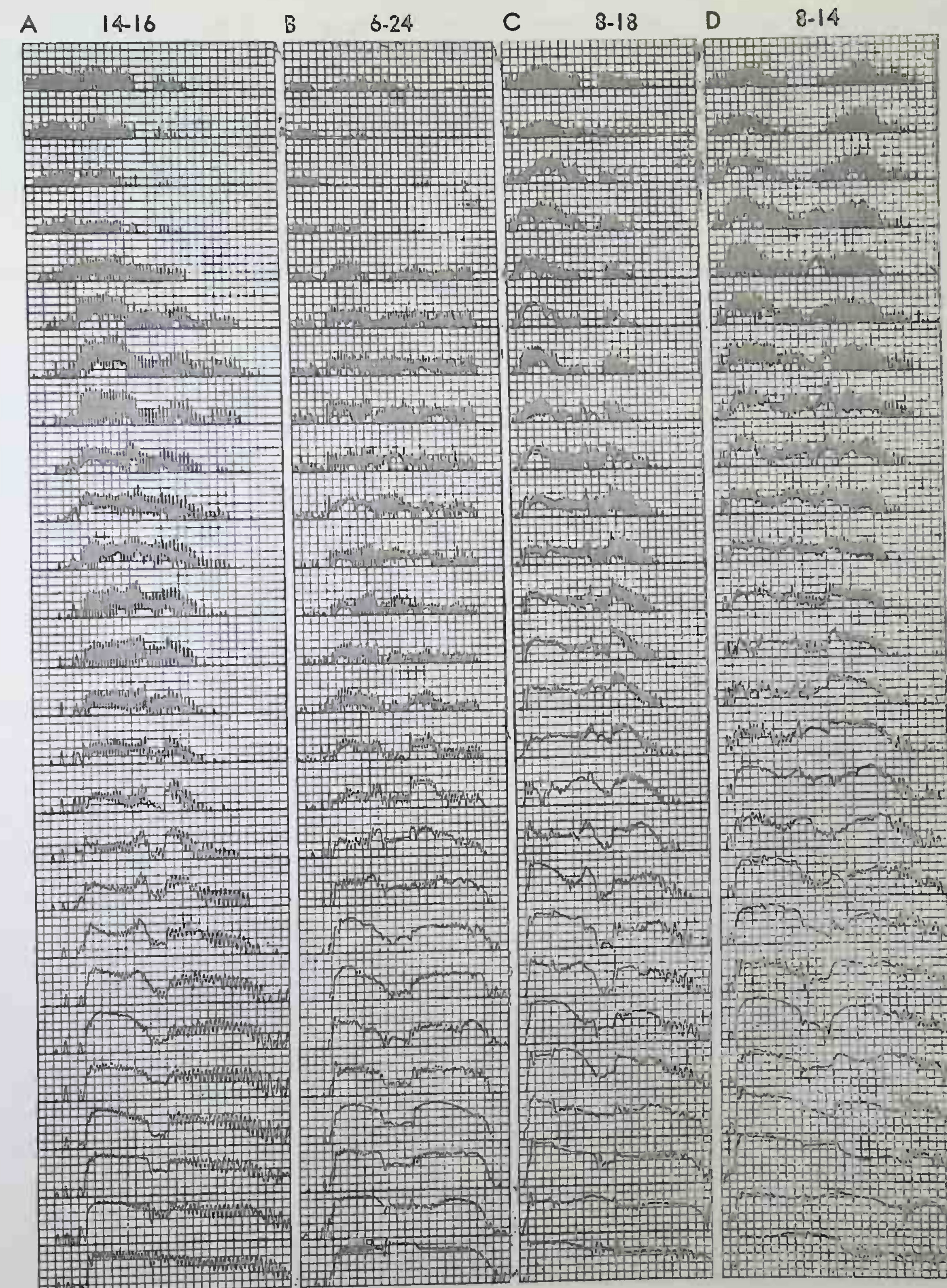

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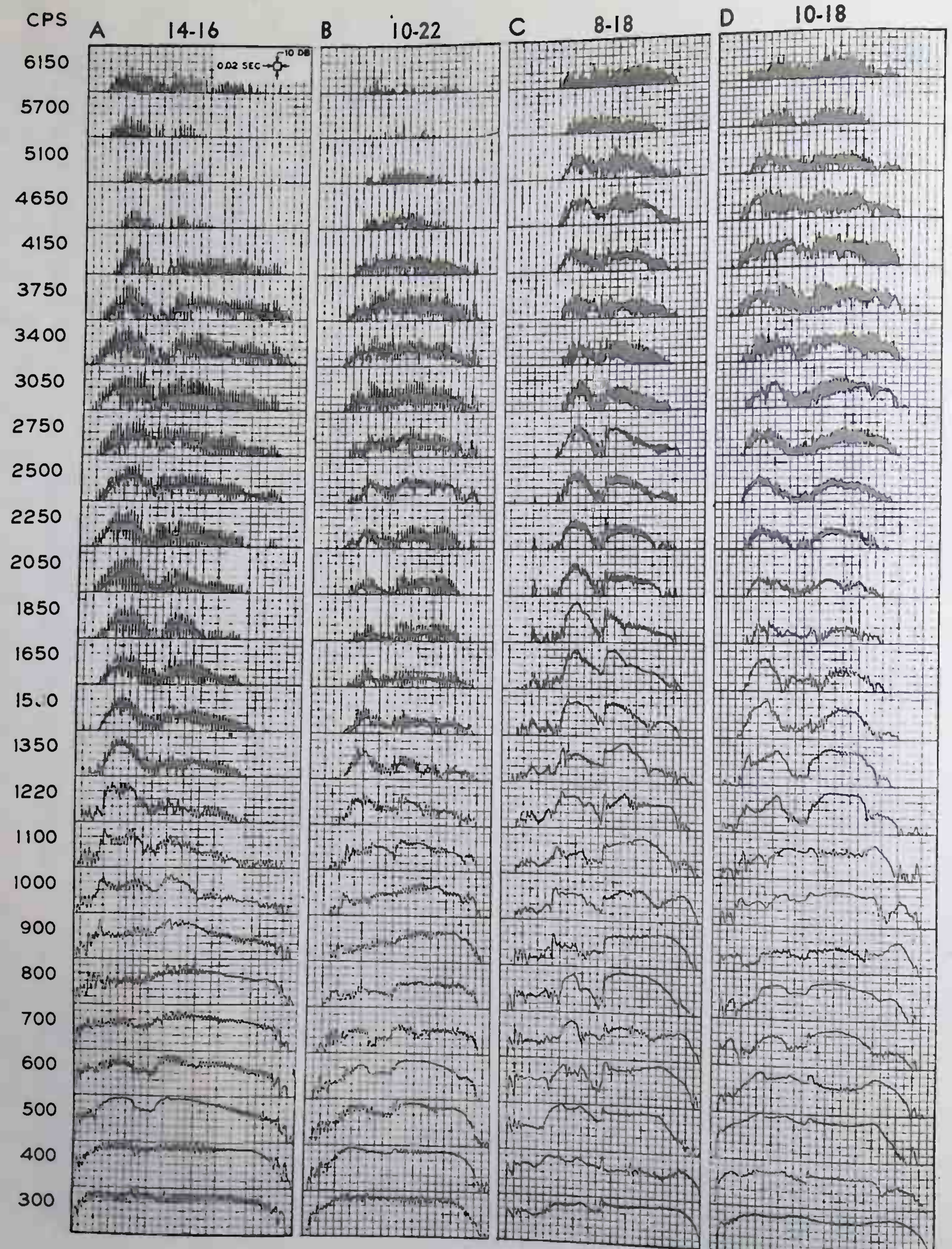
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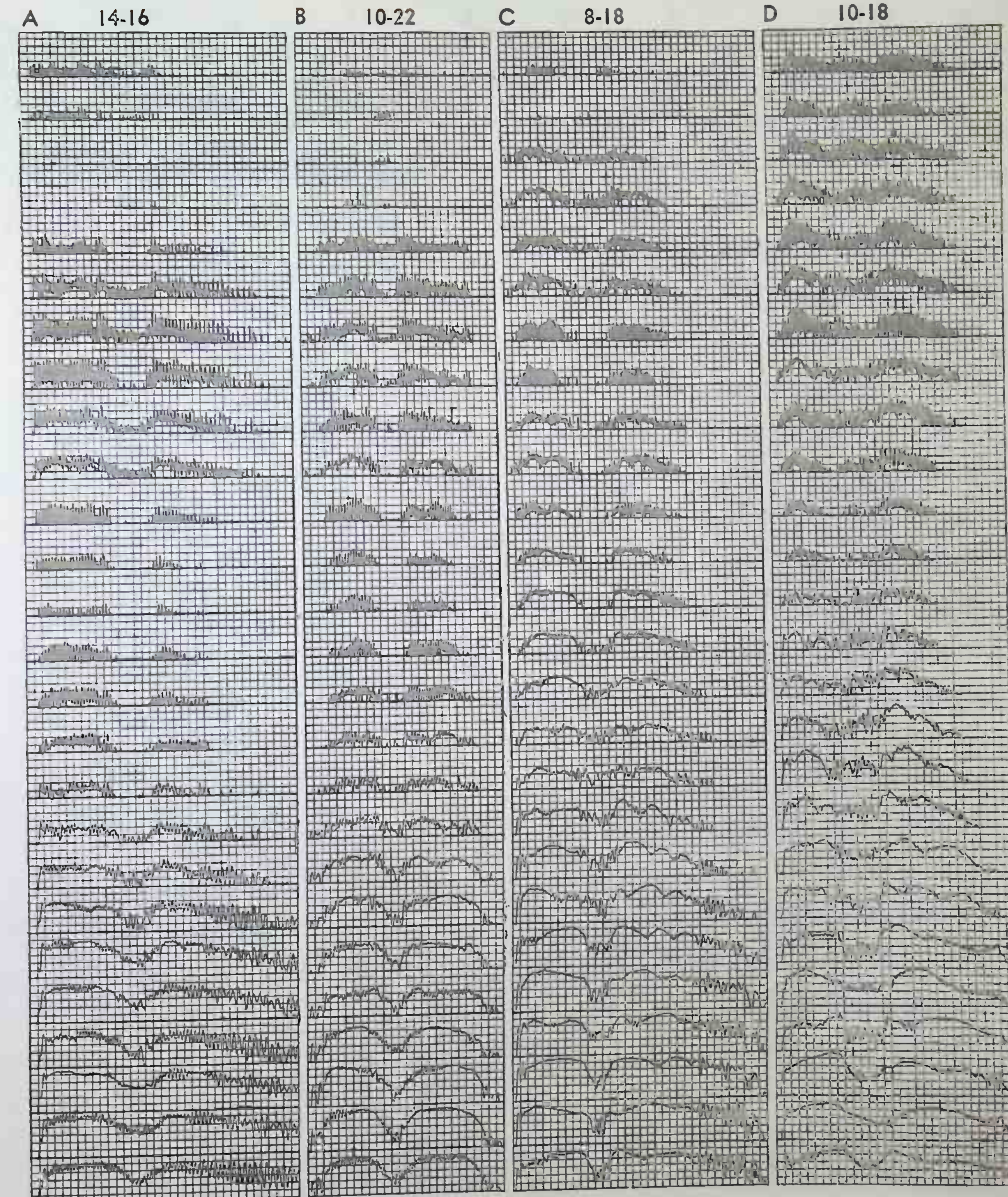
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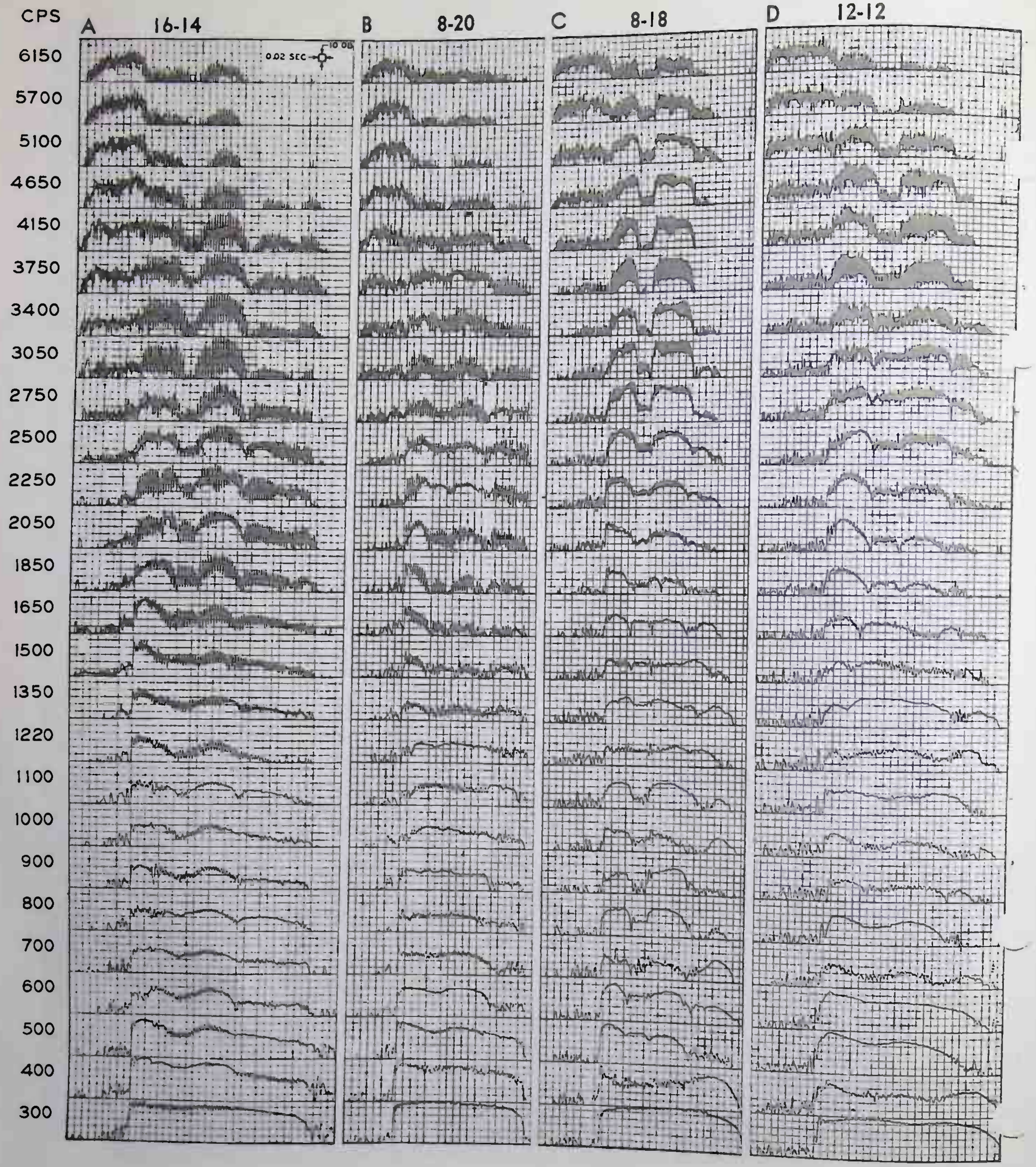

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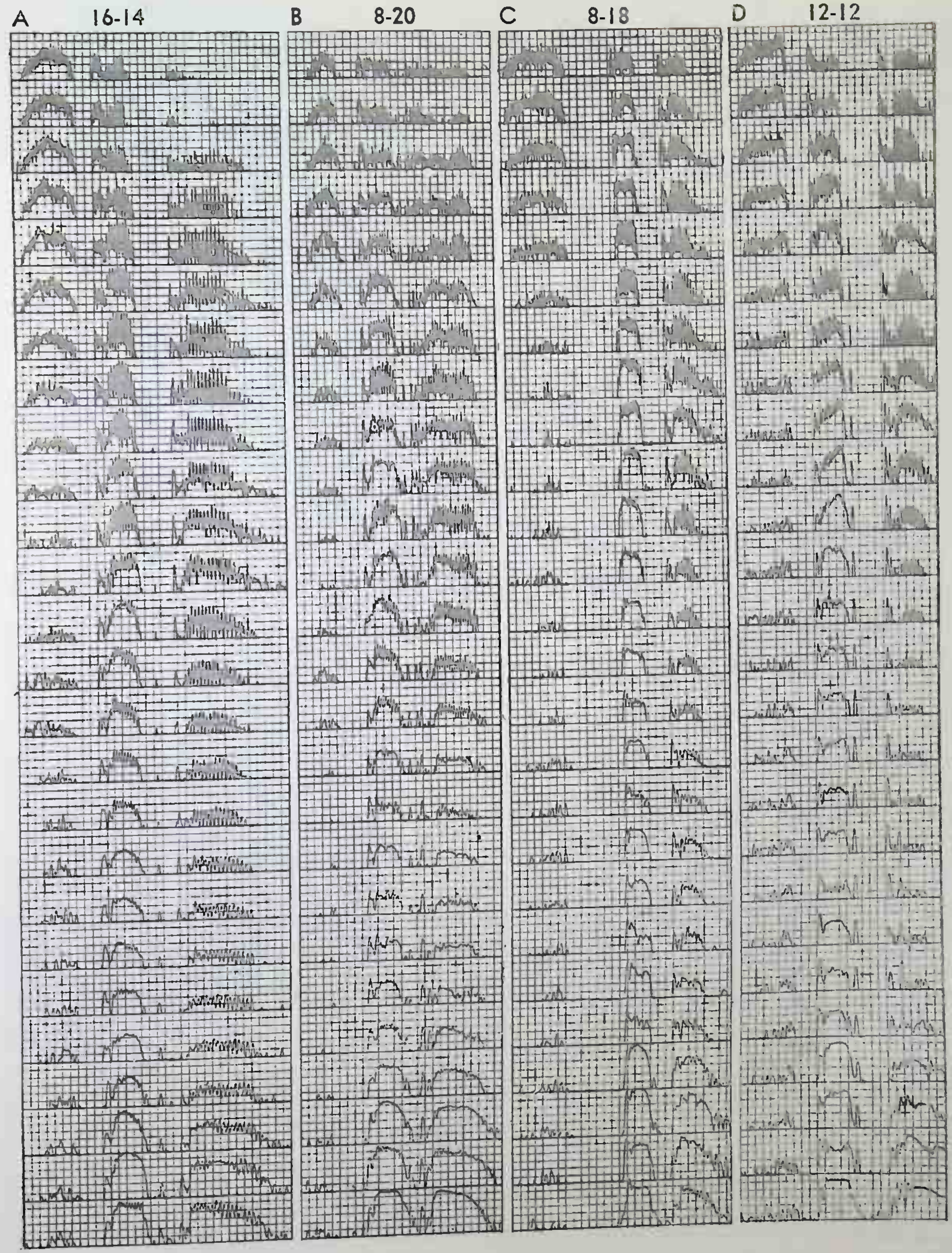

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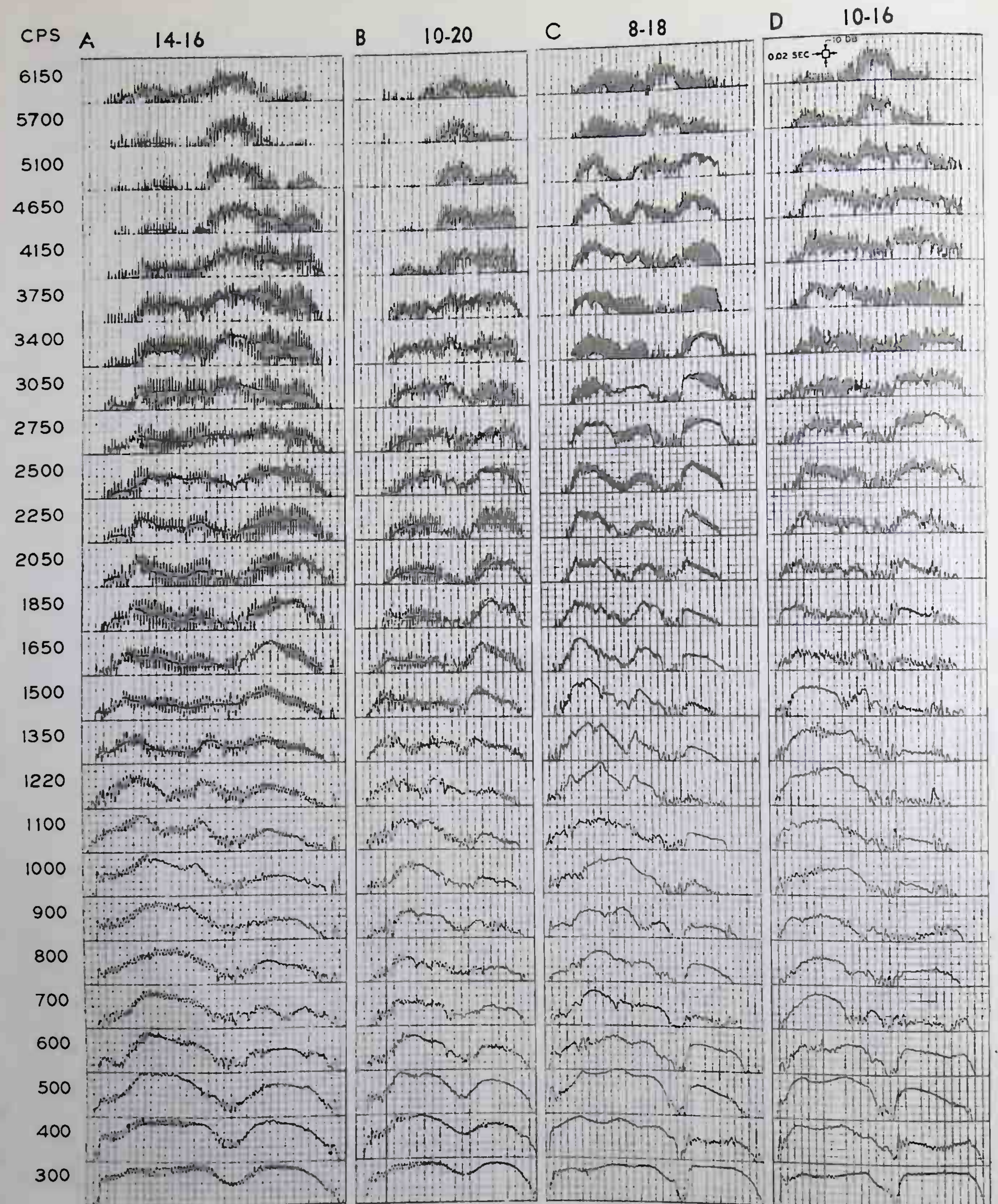

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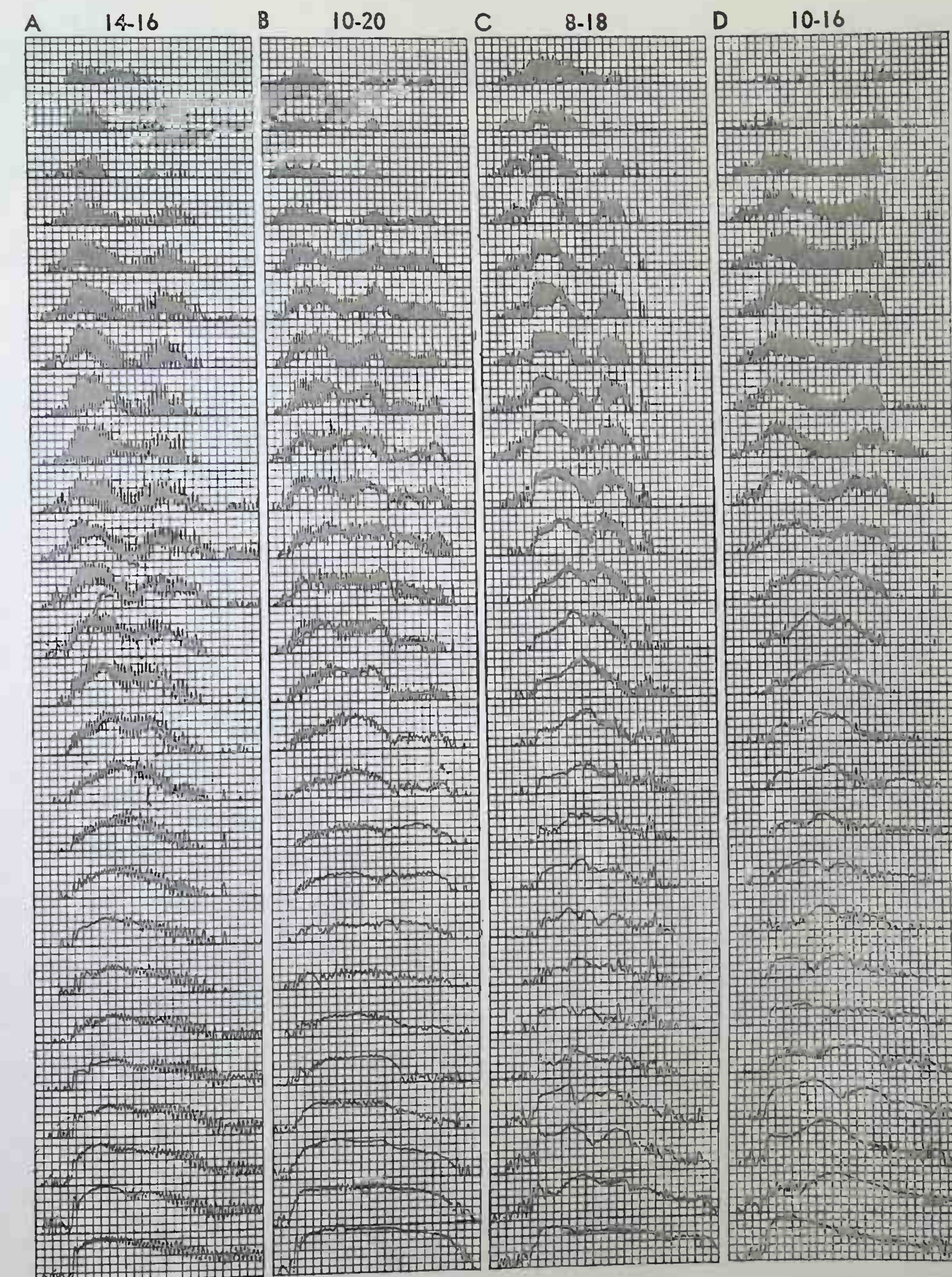

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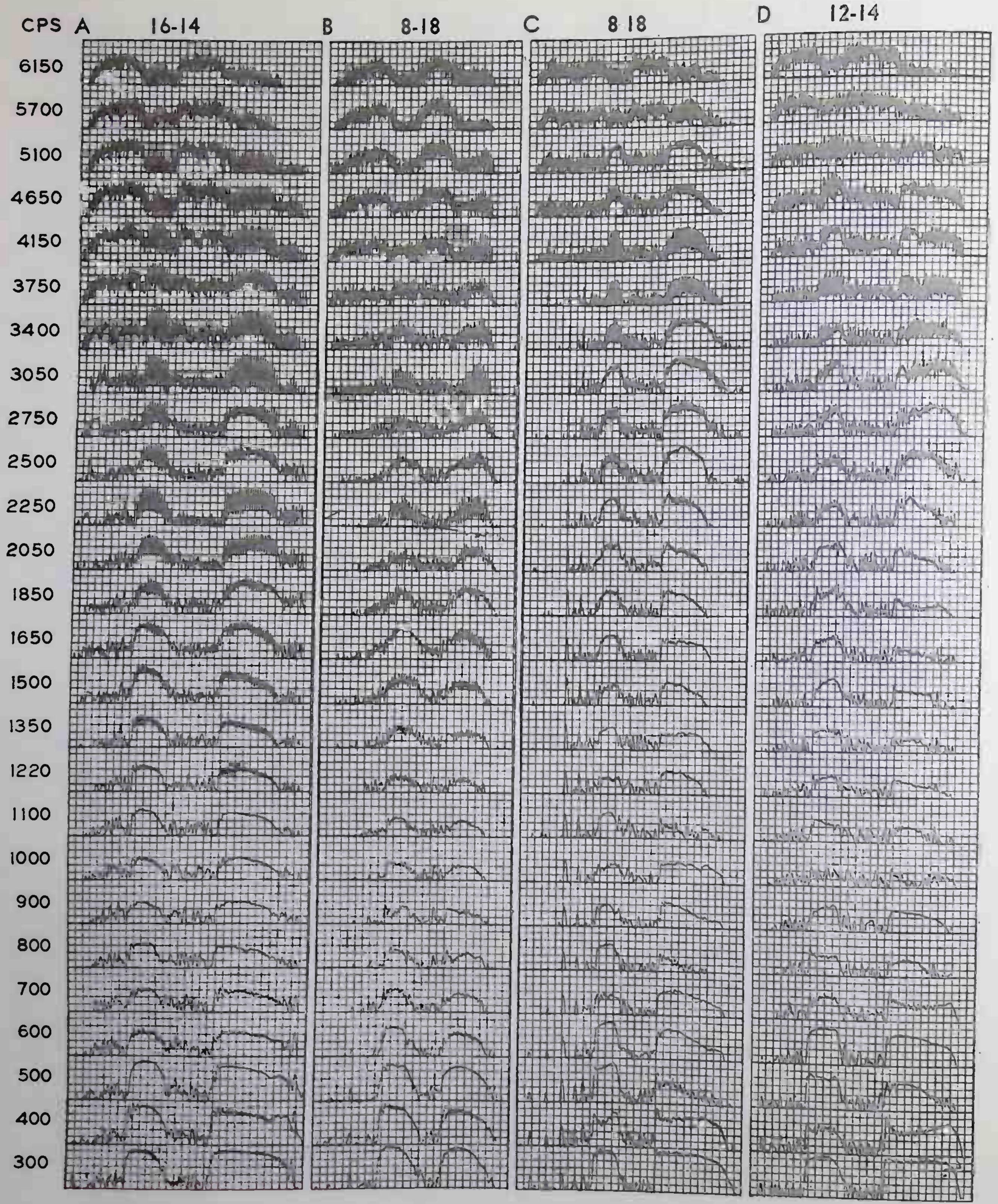

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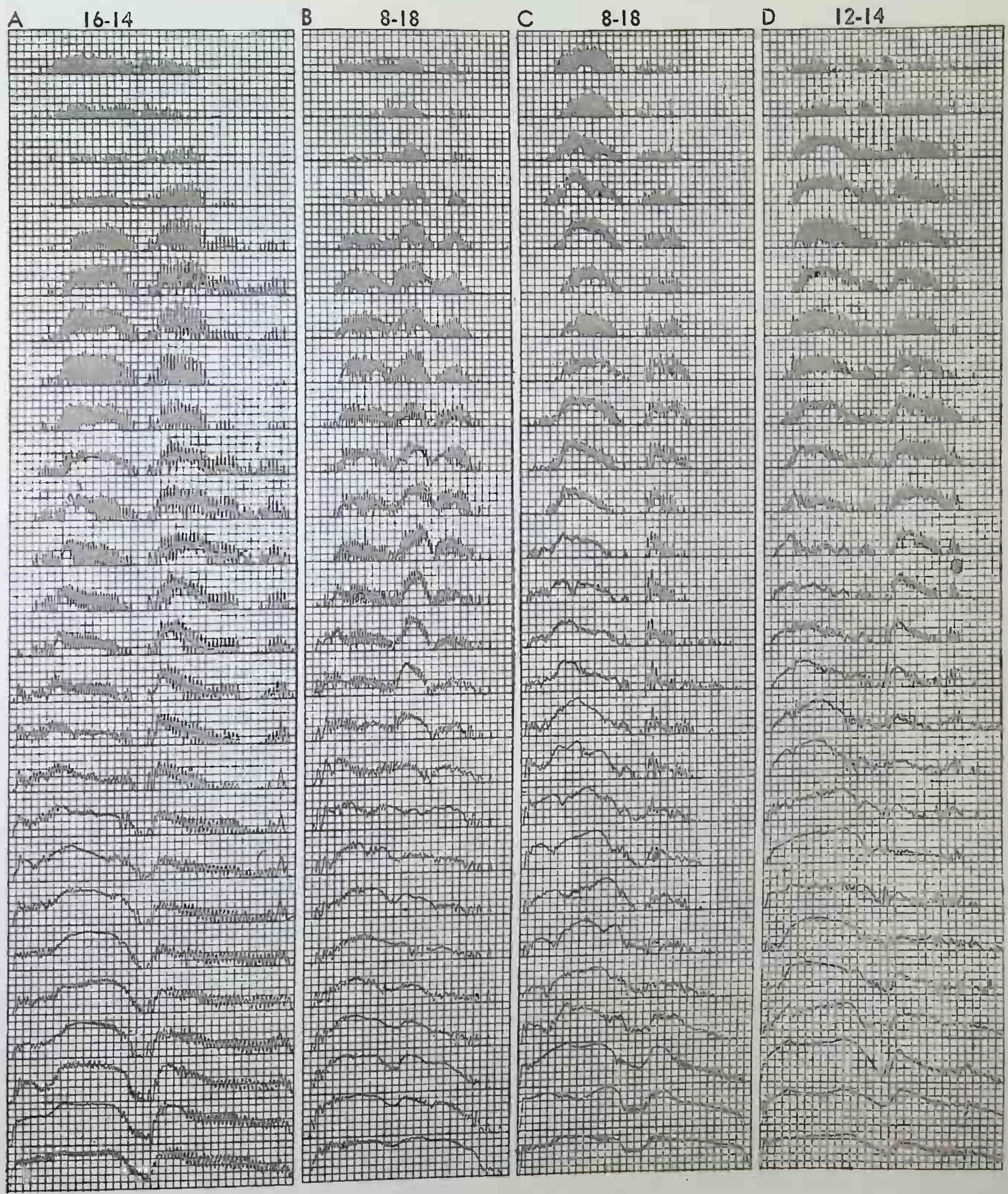

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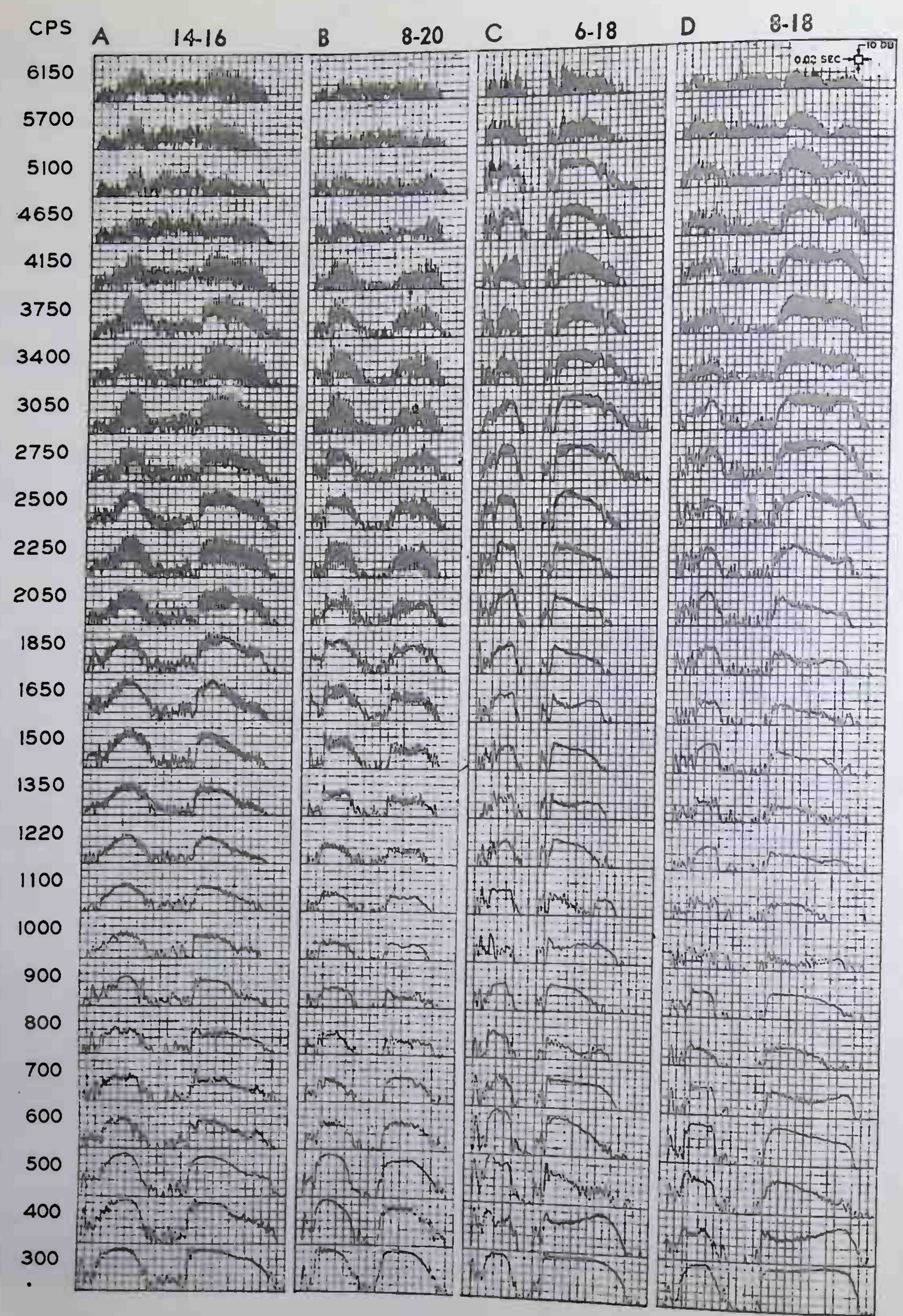

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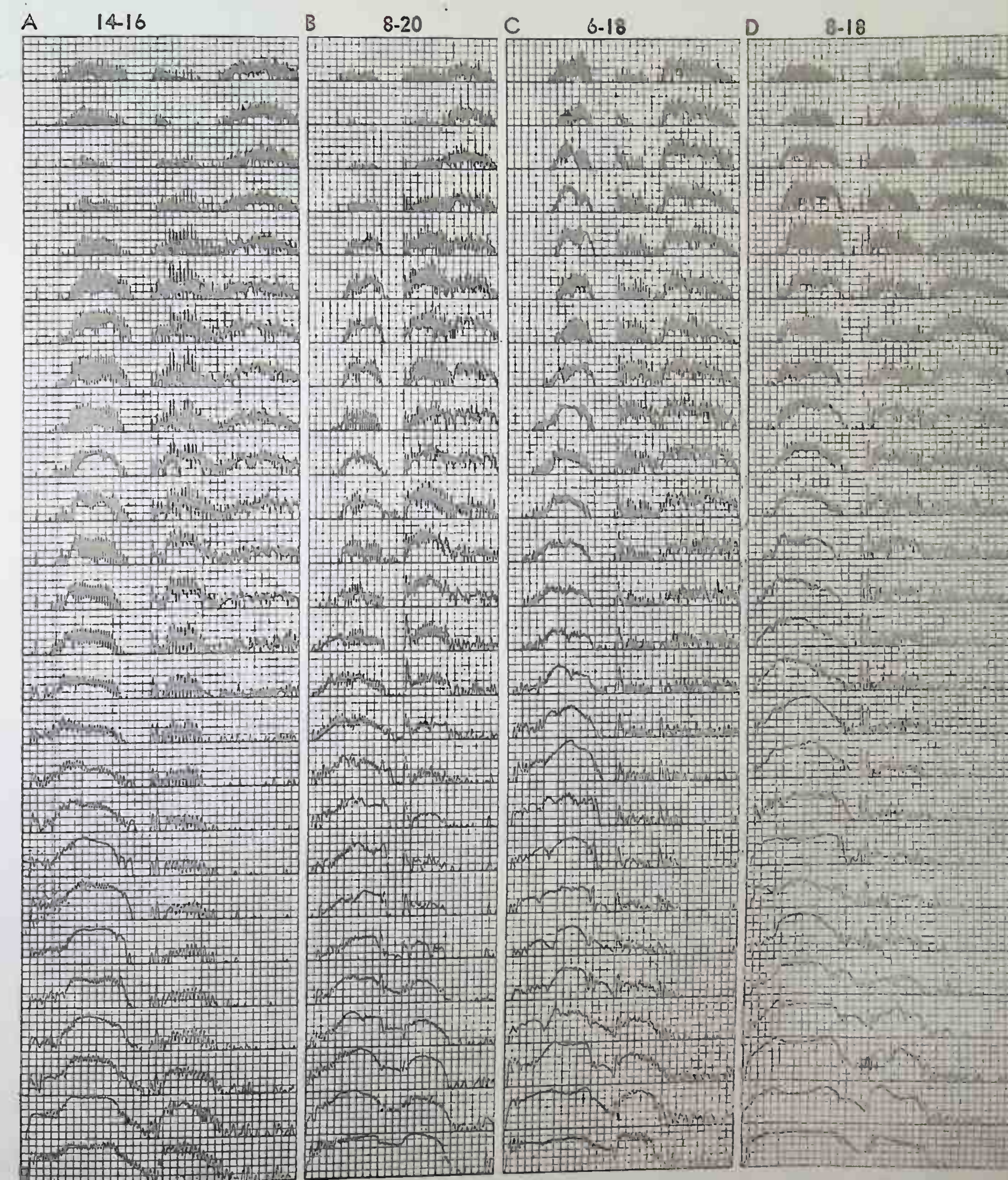

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MAY 1922

REPRINT B-6-1

THE THERMOPHONE

BY

E. C. WENTE

A DERIVATION OF FORMULAE
GIVING THE SOUND OUTPUT OF THERMOPHONES OF
BOTH THE METAL-FOIL AND THE HOT-WIRE TYPES
WITH EXPERIMENTAL VERIFICATION

REPRINT OF PAPER

Published in

THE PHYSICAL REVIEW

N. S., VOL. XIX., No. 4 APRIL, 1922

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THE THERMOPHONE.

BY EDWARD C. WENTE.

SYNOPSIS.

Acoustic Efficiency of Thermophones of the Heated Foil or Wire Type.—Theoretical formulæ are derived for the maximum value of the alternating pressure produced within the enclosure of any thermophone when a given alternating current, superposed on a direct current, is passed through the central foil or wire. The effect of certain simplifying assumptions which are made is shown to be small in practical cases. As an *experimental verification* of the formulæ, an electrostatic transmitter was calibrated for a wide range of frequencies with four thermophones which differed greatly in their physical constants, the formulæ being used to compute the pressures produced. The four calibrations thus obtained agree with each other closely and also with an independent calibration made with a pistonphone.

Methods of Calibrating Acoustical Transmitters.—In addition to the pistonphone, the thermophone is now available. The circuits used in calibrating an electrostatic transmitter are given.

Method of Measuring the Thermal Conductivity of a Gas by Using a Thermophone is Suggested.—It would have the advantage of avoiding difficulties due to convection.

IN a paper on the theory of the thermophone, Arnold and Crandall[1] have developed formulæ for the absolute value of its acoustical efficiency as a function of the frequency. While these formulæ are sufficiently accurate for most practical purposes, certain discrepancies[2] have been observed, which cannot be attributed to experimental errors. In order to use the thermophone for calibrating acoustical apparatus throughout a wide range of frequencies it has therefore been necessary to make a more detailed analysis of its action.

In this paper formulæ are derived for the efficiency of a thermophone with a heating element of metal foil, which are found to agree closely with such experimental tests as it has been possible to apply. Expressions for the efficiency of the thermophone having a wire heating element are also obtained by the same method of analysis. Only those cases are considered in which the thermophone heating element is located within an enclosure the linear dimensions of which are small in comparison with the wave length of sound. When the element is not placed within a small enclosure the thermophone is of little importance as a precision instrument. Its sound output is not only small but indefinite in most practical cases. Since the sound waves are generated by the alternate expansion and contraction of a layer of air in the immediate

[1] PHYS. REV., X, 22 (1917).

[2] PHYS. REV., X, 52 (1917).

neighborhood of the foil, the characteristics of the system are similar to those of a source of sound consisting of a diaphragm of small mass and stiffness actuated by a constant alternating force. The radiation resistance, which varies with the position of surrounding objects, may in this case be an appreciable part of the mechanical impedance of the system.

Equation of Thermal Equilibrium of an Ideal Gas.

The differential equation of heat conduction for a continuously varying temperature in a solid is well known. The corresponding equation for gases, which is made the basis of the following study of the action of the thermophone, is obtained by putting the rate of increase of heat content of an element of volume of the gas equal to the rate at which work is done upon it plus the heat received by conduction. We thus get

$$\rho c_v \frac{DT}{Dt} = -Ap\rho \frac{DV}{Dt} - K\nabla^2 T,$$

where $V = 1/\rho$ is the specific volume of the gas. Since for an ideal gas $p = \rho RT$ and $c_p - c_v = AR$, this equation reduces to

$$\rho c_p \frac{DT}{Dt} - A\frac{Dp}{Dt} - K\nabla^2 T = 0. \qquad (1)$$

The loss by radiation is small in any practical case and so has been neglected.

Thermophone with Heating Element of Metal Foil.

Pressure Produced by a Periodic Change in Temperature of a Metal Strip Within a Small Enclosure.—If the temperature of the foil varies periodically, a temperature wave will be set up in the medium surrounding the foil. At acoustic frequencies these waves are attenuated to a small fraction of their original amplitude at distances small compared with the dimensions of the foil. Hence, in this discussion these thermal waves may be regarded as plane waves; also, since the fluid velocity is small, D/Dt is very nearly equal to $\partial/\partial t$. If then the temperature of the strip is given by $\Theta_0 + \Theta_1 e^{i\omega t}$, where Θ_1 is small compared with Θ_0, and if second order effects are neglected,

$$T = T_0 + T_1 e^{i\omega t},$$
$$p = p_0 + p_1 e^{i\omega t},$$
$$\rho = \rho_0 + \rho_1 e^{i\omega t},$$

and from (1) it follows that

$$\rho_0 c_p i\omega T_1 - Ai\omega p_1 - K\frac{\partial^2 T_1}{\partial x^2} = 0. \qquad (2)$$

In the solution of this equation ρ_0 and K may be assigned the values that obtain at the temperature of the foil, since at points near the foil the gas has approximately the same temperature as the foil itself. If the dimensions of the enclosure are small compared with a wave-length of sound, p_1 is independent of x. The complete solution of (2) then is

$$T_1 = Ce^{\sqrt{2i}\alpha x} + De^{-\sqrt{2i}\alpha x} + \frac{k-1}{k}\frac{\Theta_0}{p_0}p_1,$$

in which α is set equal to $\sqrt{\dfrac{\rho_0\omega c_p}{2K}}$, and k is the ratio of the specific heats.

The boundary conditions require that for $x = \infty$, T_1 shall not be ∞ and for $x = 0$, T_1 shall equal Θ_1. Hence,

$$T_1 = \left(\Theta_1 - \frac{A\omega p_1}{2\alpha^2 K}\right)e^{-\sqrt{2i}\alpha x} + \frac{k-1}{k}\frac{\Theta_0}{p_0}p_1. \qquad (3)$$

Also

$$p_1 = \frac{2p_0 a}{T_a V_0}\int_0^{V_0/2a} T_1 dx,[1] \qquad (4)$$

where V_0 is the volume, and T_1 the mean temperature of the gas in the enclosure. Since, in all practical cases, $e^{-V_0\alpha/2a}$ is small compared with unity, it follows from (3) and (4) that

$$\Theta_1 = \frac{k-1}{k}\frac{\Theta_0}{p_0}\left[1 - \left(1 - \frac{k}{k-1}\frac{T_a}{\Theta_0}\right)\frac{V_0\alpha}{2a}(1+i)\right]p_1. \qquad (5)$$

This equation gives the periodic pressure within the enclosure in terms of the periodic temperature variation of the heating element. The problem remains to express Θ_1 in terms of quantities that can be measured directly.

Temperature of the Heating Element.—Consider the case in which the thermophone element is heated by both direct and alternating currents, *i.e.*, by a current equal to $I_0 + I_1 \cos \omega t$ amperes. If I_0 is large compared with I_1, the rate at which heat is developed in the foil is approximately equal to the real part of

$$0.239RI_0^2 + 0.478RI_0I_1e^{i\omega t} \text{ calories per second.}$$

Equating this value to the rate with which heat is lost by radiation and conduction plus the rate with which it is stored in the foil, we get for the fundamental frequency

$$04.78RI_0I_1 = 8aC\Theta_0^3\Theta_1 - 2aK\left(\frac{\partial T}{\partial x}\right)_{x=0} + a\gamma i\omega\Theta$$
$$= \Theta_1(a\gamma i\omega + 2a\alpha K\sqrt{2i} + 8aC\Theta_0^3) - \frac{Aa\omega\sqrt{2i}}{\alpha}p_1, \qquad (6)$$

[1] See Arnold and Crandall, PHYS. REV., N.S., Vol. X, 32 (1917).

where C is the radiation constant of the foil and γ is the heat capacity per unit area of foil. Reintroducing the time factor and retaining only the real part, we get finally from (5) and (6) for the pressure developed within the enclosure,

$$p_1' = \frac{0.478RI_0I_1\cos(\omega t - \Phi)}{\left[\left(GQ - RF - \frac{Aa\omega}{\alpha}\right)^2 + \left(FQ + RG - \frac{Aa\omega}{\alpha}\right)^2\right]^{1/2}}, \tag{7}$$

where

$$F = \frac{V_0\alpha T_a}{2ap_0}\left(1 - \frac{k-1}{k}\frac{\Theta_0}{T_a}\right),$$

$$G = \frac{V_0\alpha T_a}{2ap_0}\left[1 - \frac{k-1}{k}\frac{\Theta_0}{T_a}\left(1 - \frac{2a}{\alpha V_0}\right)\right],$$

$$Q = 2a\alpha K + 8aC\Theta_0^3,$$

$$R = a\gamma\omega + 2a\alpha K,$$

$$\Phi = \tan^{-1}\frac{FQ + GR - \frac{Aa\omega}{\alpha}}{GQ - RF - \frac{Aa\omega}{\alpha}}.$$

In these expressions α and K must be assigned values corresponding to the temperature of the foil. If K_0 and α_0 are the values at 0° C., then

$$K = K_0\left(\frac{\Theta_0}{273}\right)^{1/2}; \qquad \alpha = \alpha_0\left(\frac{273}{\Theta_0}\right)^{3/4}.$$

The Thermophone with Cylindrical Heating Element.

For laboratory purposes it is generally more practicable to use a thermophone with a heating element of metal foil instead of Wollaston wire as proposed by DeLange.[1] However, there are some cases in which the latter is of value as a standard source of sound, particularly in measurements on ear sensitivity. Thermophones of this type can readily be placed within the ear passage. Complications arising from stationary waves are thus almost entirely eliminated.

Pressure Within an Enclosure Containing a Cylindrical Wire of Variable Temperature.—Consider an airtight cylindrical enclosure along the axis of which is placed a wire of small diameter, having a temperature which can be represented by the expression $\Theta_0 + \Theta_1 e^{i\omega t}$. As before, the fluid velocity in the surrounding medium is small so that the relation (1) reduces to

$$K\left(\frac{\partial^2 T}{\partial r^2} + \frac{1}{r}\frac{\partial T}{\partial r}\right) + A\frac{\partial p}{\partial t} - c_p\rho\frac{\partial T}{\partial t} = 0. \tag{8}$$

[1] Proc. Royal Soc., 91A, p. 239 (1915).

For the equation corresponding to (2) we have

$$\left(\frac{\partial^2}{\partial r^2} + \frac{1}{r}\frac{\partial}{\partial r} - 2i\alpha^2\right)T_1 = -\frac{i\omega Ap_1}{K}. \tag{9}$$

If, as before, α is assumed to be constant, the complete solution of (9) is

$$T_1 = AJ_0(\sqrt{-2i\alpha}r) + BH_0(\sqrt{-2i\alpha}r) + \frac{k-1}{k}\frac{\Theta_0}{p_0}p_1.$$[1]

The boundary conditions require that, as r approaches ∞, T_1 shall not approach ∞, and when r is equal to a, the radius of the wire, T_1 shall be equal to Θ_1. Hence

$$T_1 = \frac{\left(\Theta_1 - \frac{k-1}{k}\Theta_0\frac{p_1}{p_0}\right)}{H_0^{(2)}(\sqrt{-2i\alpha}a)}H_0^{(2)}(\sqrt{-2i\alpha}r) + \frac{k-1}{k}\Theta_0\frac{p_1}{p_0} \tag{10}$$

corresponding to (4) we have in this case the relation:

$$p_1 = \frac{p_0 l}{T_a V_0}\int_a^R T_1 2\pi r\,dr, \tag{11}$$

where R is the radius of the cylindrical enclosure, and l the length of the wire. In any practical type of thermal receiver R is large compared with a, and $aH_1^{(2)}(\sqrt{-2i\alpha}a)$ is large compared with $RH_1^{(2)}(\sqrt{-2i\alpha}R)$, so that to a close approximation we have from (10) and (11)

$$p_1 = \frac{2p_0\pi al\left(\frac{k-1}{k}\frac{\Theta_0}{p_0}p_1 - \Theta_1\right)H_1^{(2)}(\sqrt{-2i\alpha}a)}{\sqrt{-2i\alpha}V_0\left(T_a - \frac{k-1}{k}\Theta_0\right)H_0^{(2)}(\sqrt{-2i\alpha}a)}. \tag{12}$$

Temperature of the Heating Element.—Setting the rate with which heat is supplied to the wire equal to the rate with which heat is lost by conduction and radiation plus the rate with which heat is stored, we obtain the following relation

$$\begin{aligned} 0.478RI_0I_1 &= l\left[-2\pi aK\left(\frac{\partial T}{\partial r}\right)_{r=a} + 8\pi aC\Theta_0^3\Theta_1 + \gamma' i\omega\right] \\ &= l\left[\gamma' i\omega\Theta_1 + 2K\pi\frac{\sqrt{-2i\alpha}aH_1^{(2)}(\sqrt{-2i\alpha}a)}{H_0^{(2)}(\sqrt{-2i\alpha}a)}\left(\Theta_1 - \frac{\omega Ap_1}{2\alpha^2 K}\right)\right. \\ &\qquad \left. + 8\pi aC\Theta_0^3\Theta_1\right], \end{aligned} \tag{13}$$

[1] The notation here used for the Bessel functions is the same as given in Jahnke and Emde, "Functionen-tafeln," p. 90 ff.

where γ' is the heat capacity per unit length of the wire. Let

$$\frac{H_0^{(2)}(\sqrt{-2i\alpha a})}{H_1^{(2)}(\sqrt{-2i\alpha a})} = M + iN.$$

We then get from (12) and (13)

$$0.478RI_0I_1 = \left[(\gamma' i\omega + 8\pi aC\Theta_0^3)\left(\frac{k-1}{k}\frac{\Theta_0}{p_0}l\right.\right.$$
$$\left. - \frac{(1-i)\left(T_a - \frac{k-1}{k}\Theta_0\right)V_0}{2p_0\pi a}(M+iN)\right)$$
$$\left. + \frac{2i\alpha^2KV_0\left(T_a - \frac{k-1}{k}\Theta_0\right)}{p_0}\right]p_1. \tag{14}$$

In general, $8\pi aC\Theta_0^3$ is small compared with $\gamma'\omega$. For example, in the case of a thermophone having a wire heating element, 0.0003 cm. in diameter, for a frequency of 25 cycles per second and $\Theta_0 = 800°$ absolute, the error, introduced by neglecting $8\pi aC\Theta_0^3$ in (14), even for these extreme conditions, is only 4 per cent. If then $8\pi aC\Theta_0^3$ is neglected, we obtain finally for the real part of p_1, the alternating pressure developed within the enclosure,

$$p_1' = \frac{0.956\pi ap_0}{\left(T_a - \frac{k-1}{k}\Theta_0\right)\alpha V_0\omega\gamma'}\,\frac{RI_0I_1\cos(\omega t - \Phi)}{\left[(N-M)^2 + \left(\frac{k}{k-1}\frac{2\pi aAp_0}{\alpha\Theta_0\gamma'} + \frac{2\pi al}{V_0\alpha\left(\frac{k}{k-1}\frac{T_a}{\Theta_0} - 1\right)} - (M+N)\right)^2\right]^{1/2}}, \tag{15}$$

where

$$\Phi = \tan^{-1}\left[\frac{\frac{k}{k-1}\cdot\frac{2\pi aAp_0}{\alpha\Theta_0\gamma'} + \frac{2\pi al}{V_0\alpha\left(\frac{k}{k-1}\frac{T_a}{\Theta_0} - 1\right)} - (M+N)}{N-M}\right].$$

In the calculation of the values of $N - M$ and $M + N$ for small values of αa, it is convenient to use the following approximation formulæ,

$$N - M = 2\alpha a\left[\log_e \frac{\alpha a}{\sqrt{2}} + 0.577\right]; \qquad N + M = \frac{\alpha a}{2}.$$

These formulæ are correct to within a few per cent. for values of αa even as large as 0.15. For larger values of αa, M and N may be calculated from the tables given in Jahnke and Emde's "Functionentafeln" pp. 139–140. $H_\nu^{(2)}(\sqrt{-i}X)$ is the conjugate complex quantity of $H_\nu^{(1)}(\sqrt{i}X)$.

Values for the maximum pressure developed have been calculated by the above formulæ for thermophones having the following constants.

I. Heating element of metal foil: volume of the enclosure = 14 cu. cm., thickness of the gold foil = 0.79×10^{-5} cm., area of the gold foil = 5.5 sq. cm., $\Theta_0 = 335°$ absolute, $T_a = 300°$ absolute.

II. Heating element of Wollaston wire: volume of the enclosure = 1 cu. cm., diameter of the wire = 0.0003 cm., length of the wire = 1 cm., T_a = 300° absolute, $\Theta_0 = 335°$ absolute.

Fig. 1.

Frequency characteristics of thermophones.

A. Cylindrical heating element.

B. Flat-strip heating element.

In Fig. 1 the products of pressure and the 3/2 power of the frequency are plotted as a function of frequency. The curves thus show the amount of deviation from a simple 3/2 power relation. It appears that as telephone receivers the two types of instruments produce approximately the same amount of distortion and so neither possesses an advantage in this regard.

In the preceding analysis the heating element was assumed to lie along the axis of a cylindrical enclosure. But since the temperature waves emanating from the wire have a large attenuation and a short wave length, the equations apply without alteration to any case in which the wire is located near the central part of the enclosure.

Further Considerations of the Formulæ.

In the preceding discussion the following assumptions have been made: The walls of the enclosure are non-conductors of heat, the temperature of the foil has the same value throughout its entire length, and at the surface of the foil the gas has the same temperature as the foil itself. Although for most practical cases these assumptions do not introduce inaccuracies of any consequence into the formulæ, the order of magnitude of the errors will here be considered.

If the walls are perfect conductors, as is approximately true in most cases, it may be shown that a first order correction is obtained by multiplying (7) by

$$\sqrt{1-\frac{S}{V_0\alpha}+\frac{S^2}{2V_0^2\alpha^2}},$$

where S is the area of the walls of the enclosure. This factor is in general nearly unity.

In practice, the ends of the foil are held between comparatively massive clamps, so that almost no temperature variation at these points is produced by the alternating current. To take account of this fact the values of p_1' as given by equations (7) and (15) should be multiplied by

$$\sqrt{1-\frac{2}{\alpha' l}+\frac{2}{(\alpha' l)^2}},$$

where l is the length of the element and α' is the value of α for the heating element. It may be shown that this factor gives the correction to the first order and is in general also nearly equal to unity.

When heat flows from a solid wall to a gas, the temperature corresponding to the velocity with which a molecule leaves the wall after a single impact will not be the same as the temperature of the wall. This fact is equivalent to a temperature drop between the solid and the gas.[1] This temperature drop at the surface of the foil is given by

$$\Delta\Theta = -\gamma\left(\frac{\partial T}{dx}\right)_{x\doteq 0},$$

where γ is a constant depending on the gas and its mean free path and on the condition of the surface of the solid wall. For hydrogen and a

[1] Kundt and Warburg, Pogg. Ann., 156, p. 177 (1875).

smooth solid surface on platinum it is equal to 7.25λ,[1] where λ is the mean free path of the molecules of the gas. For normal atmospheric conditions we find in this case to a first approximation $\Delta\Theta_1 = 3 \times 10^{-4} f^{1/2}\Theta_1$. For a frequency as high as 10,000 cycles this is only $0.027\Theta_1$ or less than 3 per cent. of the maximum value of the periodic temperature of the foil. For all practical purposes, therefore, the temperature drop at the surface of the foil may be disregarded.

Experimental Verification of the Theory of the Thermophone with Heating Element of Metal Foil.

(*a*) *Comparison of Thermophones having Different Physical Constants.*—There appears to be no way in which the pressure produced by a thermophone within an enclosure can be measured over its useful frequency range. The validity of equation (7), therefore, cannot be tested in a direct way. However, the alternating pressure exerted on the diaphragm of an electrostatic transmitter by thermophones having different physical constants may be compared. Whatever the sensitivity of an electrostatic transmitter as a function of frequency may be, it does not vary appreciably with time. Thus it is possible to determine at a number of fixed frequencies whether the alternating pressure varies with the different physical constants in accordance with equation (7). Experiments to this end have been carried out for a wide range of frequencies.

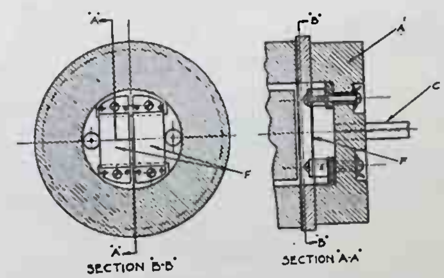

Fig. 2.

Thermophone in place for calibrating an electrostatic transmitter.

The thermophone was placed in front of the electrostatic transmitter in the manner shown in Fig. 2. The face of the transmitter together with the brass block, A, form an enclosure in which the heating elements, F, are centrally located. The enclosure is airtight except for the two equal capillary tubes, C. By means of these tubes gas is passed at a slow rate through the enclosure. To keep the pressure at atmospheric value, the pressure head on the intake tube is made equal to the suction

[1] S. Weber, A.d.P., 54, 439 (1917).

head on the outlet tube. These capillary tubes have a high impedance to the transmission of sound waves so that they do not affect the magnitude of the alternating pressure within the enclosure.

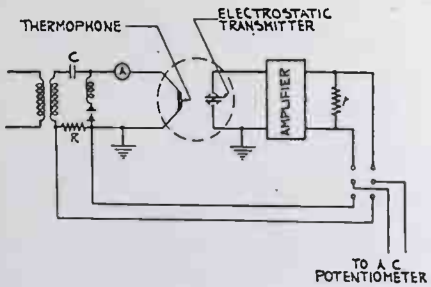

Fig. 3.

Circuit for calibrating an electrostatic transmitter with a thermophone.

The circuit used in the measurements is shown in Fig. 3. The ratio of the alternating potential drop across r and R was determined with an *a.c.* potentiometer.[1] This ratio multiplied by R and divided by the amplification of the amplifier gives the ratio of the voltage generated by the electrostatic transmitter to the alternating current passing through the thermophone heating element. The voltage per unit of pressure was then calculated by equation (7). Measurements were made with platinum foil, 6.81×10^{-5} cm. $\times$ 1.0 cm. $\times$ 5.35 cm., within an enclosure of 25.25 cu. cm. capacity, and with gold foil, 7.85×10^{-6} cm. $\times$ 1.0 cm. $\times$ 5.5 cm., within an enclosure of 13.9 cu. cm. capacity, both with hydrogen and with air within the enclosure. The conductivity of air was taken as 5.68×10^{-5} and of hydrogen, 41.6×10^{-3} c.g.s. units.

The values of voltage per dyne pressure as obtained with these thermophones for a particular electrostatic transmitter are given in the following table.

TABLE I.

Frequency	Gold Foil in Hydrogen.	Gold Foil in Air.	Platinum in Air.	Platinum in Hydrogen.
Cycles per sec.				
60	6.75		6.75	
300	4.85	5.00	4.82	
600	4.40	4.45	4.47	4.47
1000	4.20	4.30	4.45	4.30
2000	4.06	4.20		4.25
4000	4.40			4.55

[1] E. C. Wente, A Vacuum-Tube Alternating Current Potentiometer, Journal of the A.I.E.E., 40, p. 900, December, 1921.

It is seen from this table that the different thermophones yield practically the same results. Observations were not extended over the whole frequency range for each thermophone, since equation (7) applies only for a limited region. The frequency must be lower than the resonant frequency of the enclosure, and the wave length of the temperature wave emanating from the foil must be small compared with the dimension of the enclosure.

At the lower frequencies the predominating quantity in the denominator of equation (7) for gold in hydrogen is $2\alpha K$, while for platinum in air it is $\gamma\omega$. The tests, therefore, cover extreme conditions. Hence the fact that the values obtained with the different thermophones lie very near together is evidence for the correctness of equation (7), on the basis of which they were determined.

(*b*) *Comparison with a Pistonphone.*—If at some part of the frequency range it is possible to determine by an independent method the sensitivity of the electrostatic transmitter, a more direct experimental check of formula (7) can be obtained. Measurements were made on the electrostatic transmitter similar to those just described but with the thermophone replaced by a pistonphone. The construction of this apparatus is shown in Fig. 4. This pistonphone is similar to that described in a

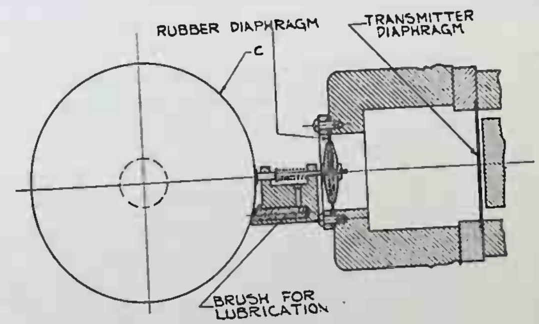

Fig. 4.

Use of pistonphone for calibrating an electrostatic transmitter.

former paper,[1] except that a cam, C, is used, which is cut so that with uniform rotation, a simple harmonic motion of 2 cycles per revolution is imparted to the piston. The pressure exerted on the diaphragm is given by

$$\frac{kp_0\sigma\xi\cos\omega t}{V_0}\sqrt{1-\frac{S}{V_0\alpha}+\frac{S^2}{2V_0^2\alpha^2}}, \qquad 16)$$

[1] E. C. Wente, PHYS. REV., X, p. 48 (1917).

where σ is the effective area of the face of the piston,
S is the area of the walls of the enclosure,
$\xi \cos \omega t$ is the displacement of the piston.

The last factor in (16) takes account of the fact that, due to the heat conduction of the walls, the compression does not take place quite adiabatically.

A diagram of the circuit used in this test is shown in Fig. 5. If the electrostatic capacities of the transmitter and voltmeter are known, the open-circuit voltage of the transmitter may be determined from the voltmeter readings. Measurements were made for frequencies ranging from 10 to 200 cycles per second. The open-circuit voltages per dyne pressure are plotted in Fig. 6 together with values obtained for the same electrostatic transmitter with a gold leaf-hydrogen thermophone. The points are seen to lie very near together. This fact shows that for this frequency range the absolute value of the pressure produced by the thermophone as a function of frequency is correctly given by equation (7). At the lower frequencies the pressure produced by the thermophone varies with the different factors entering into the

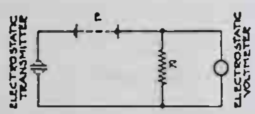

Fig. 5.

Circuit for calibrating electrostatic transmitter with pistonphone.

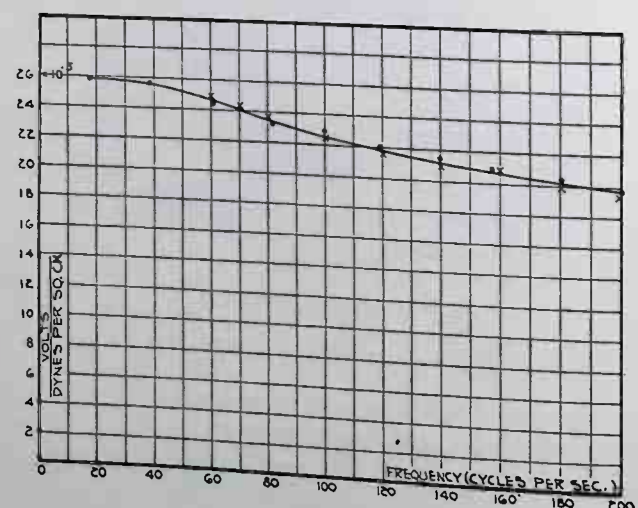

Fig. 6.

Efficiency of electrostatic transmitter.
• Values obtained with the pistonphone.
× Values obtained with the thermophone.

expression (7) more than at the higher frequencies. Therefore, it is reasonable to assume that this expression also gives correct values at higher frequencies.

The Use of the Thermophone for Measurements of the Thermal Conductivity of Gases.

One cause of uncertainty in the measurements with the thermophone is the fact that the thermal conductivity of gases is not well known. For example, in the case of hydrogen good authority can be found for values which differ from each other as much as 15 per cent. The thermophone suggests itself as a possible method for determining the conductivity of a gas. The principal source of error in conductivity measurements as heretofore carried out lies in the transport of heat by convection currents. In the case of the thermophone this source of inaccuracy is entirely eliminated.

Various arrangements of the apparatus for carrying out thermal conductivity measurements with the thermophone could be suggested. Some independent method of measuring an alternating pressure or producing one of known magnitude is required. Apparatus of the pistonphone type is perhaps the simplest arrangement that could be devised for the purpose.

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THE PRINCIPLES OF THE LIGHT VALVE

BY

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2

A DISCUSSION OF THE ENGINEERING FACTORS IN THE DESIGN AND USE OF LIGHT VALVES AND A DESCRIPTION OF AN IMPROVED TYPE

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THE PRINCIPLES OF THE LIGHT VALVE*

T. E. SHEA, W. HERRIOTT, AND W. R. GOEHNER**

Summary.—*The light valve has been used very widely as the modulating device in systems of film sound recording. In this paper the principles of operation of the light valve are discussed, and those engineering factors which prescribe limitations on performance and indicate operating advantages are described in detail. The type of distortion which results when a light valve is overloaded is depicted both for single-plane and two-plane valves. Finally, a new type of light valve having advantages from the standpoints of weight, size, and stability of operation is described.*

I. Introduction

The light valve, as a sound recording instrument, has seen very wide use during the past three years. It has undoubtedly been used for more recording and re-recording in the motion picture industry during that time than have all other types of light modulating devices combined. The extensive experience acquired with its use in studio, location, and newsreel recording has shown it to be a rugged and efficient instrument capable of making sound records of excellent quality.

During this time extensive studies have been carried on to perfect the light valve. For instance, as the producers have become better acquainted with sound recording systems and have been able to utilize them more nearly to their full capabilities, the quality of sound recording has improved, and it has been necessary to improve various elements of recording systems in order to extend their range of operation and to reduce to a minimum production delays due to recording difficulties. How great a change in recording conditions has taken place during the last three years may be seen by considering, as a case in point, the early difficulties encountered with outdoor recording on location in contrast with the smoothness and regularity with which recording equipment is now operated under similar circumstances.

As often happens with an instrument which has been perfected to a

* Presented at the Fall, 1931, Meeting at Swampscott, Mass.

** Bell Telephone Laboratories, Inc., New York, N. Y.

high degree prior to commercial use, the improvements which have resulted from the studies mentioned above, though numerous, have not been fundamental. They represent rather an aggregation of minor improvements which, taken as a body, constitute an important advance. A new type of light valve, described later in this paper, does, however, represent fundamental advances.

Although various earlier attempts to construct light gates of variable aperture for sound recording had been made, the first

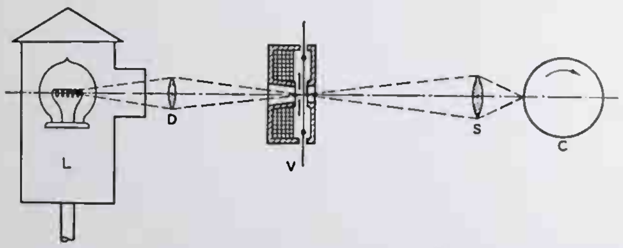

FIG. 1. The receiving end (optical system) of a picture transmission system; *V*, the light source; *D* and *S*, condensing and objective lenses; *C*, the moving film.

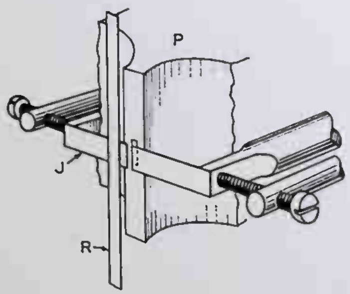

FIG. 2. The single-ribbon type of light valve; *R*, the ribbon; *J*, the aperture jaws.

practical form of light valve was developed by E. C. Wente in 1922. Subsequently, the light valve has been used in the "single-ribbon" form since 1924 for the regular commercial transmission of pictures over telephone circuits.[1]

In this development, the picture is broken into a series of long, narrow sections, similar to sound tracks, which when illuminated are scanned by a slit and photoelectric cell. The electric currents thus generated are amplified and sent over telephone wires. Processes of frequency modulation are used which need not be described here. At the receiving end these currents modulate a light valve which varies the exposure of a moving film, and provide the proper latent image for the re-creation of the transmitted picture. Fig. 1 shows the receiving end optical system of a picture transmission system, *V* being the light valve, *L* the light source, *D* and *S* the condensing and objective lenses, and *C* the moving film. Fig. 2

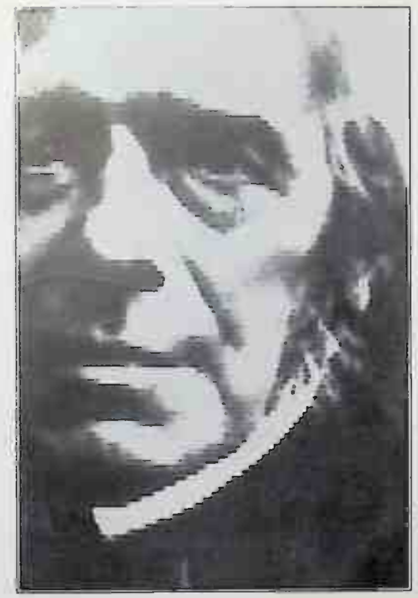

FIG. 3. Enlarged portion of a transmitted picture of the variable density type.

shows the single-ribbon type of light valve employed, the ribbon *R* vibrating in front of the gap between the aperture jaws *J*. Fig. 3 shows an enlarged portion of a transmitted picture of the variable density type; the similarity of its horizontal sections to sound tracks is obvious.

The conditions of use of light valves in sound recording are rather different from those of picture transmission, and a consideration of some of them led to the choice of the double-ribbon type of valve for this field. In general, it was recognized that in the latter field the proposed conditions of operation were more severe; and it was believed that, if the duty of modulating could be carried out by two ribbons instead of one, the sensitivity of the valve would be increased, the internal temperature rise due to conductor heating reduced, and tones freer from spurious harmonics obtained. Subsequent experience has verified these suppositions.

This is not to say that the single-ribbon valve is not suited to sound recording, but that under present conditions, at least, it will do so only at a disadvantage from several fundamental design standpoints. In what follows, where a comparison between the two types is made, an endeavor will be made to separate these factors out of any specific design and consider them on a general basis, so that the facts involved may not be clouded.

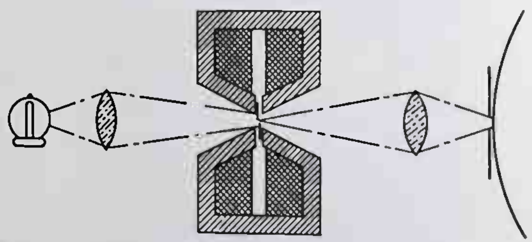

Fig. 4. An early double-ribbon valve whose moving condensers lie in different planes.

It should be kept in mind, however, in order to dissipate a prevalent but erroneous belief, that *either the single- or the double-ribbon valve may have two forms:* (*a*) *one in which light barriers or gates adjacent to the aperture are in the same plane, and* (*b*) *one in which these barriers are in different planes.* Thus, in the general form of either type of valve, excessive modulation does not lead to "light valve clash" but merely to a cutting off of the peaks of one side of the signal wave. Indeed, in one of the earliest discussions[2] of light valve operation there is described (Fig. 4) a double-ribbon valve whose moving conductors lie in different planes.

In the use of the light valve as an optical rectifier, valves of the two-plane type are requisite. In the use of a light valve as a simple modulator, the choice between the *one-plane* and *two-plane* types is to be determined by efficiency, quality, and maintenance considerations. These will be discussed later.

The general method of employing the light valve in variable density sound recording has been described at length by MacKenzie.[3] The photographic and recording technic outlined by him is sufficiently representative of present-day procedure to be assumed in what follows. The light valve itself has undergone changes in form which will be described. It is important to note in this connection, however, the specific changes in recording technic involved (1) in the use of a 1.0-mil normal aperture (instead of 2.0 mils) imaged on the film (Fig. 5) as an exposure beam 0.5 mil wide, and (2) in the use of noise reduction equipment such as that described by Silent.[4]

Either the single- or double-ribbon valve may be used not only for variable density, but for variable width recording. In the former case, the direction of the ribbons is transverse to the film; in the latter case, lengthwise with respect to it. The light valve has been used for variable width work in picture transmission. Inasmuch as all commercial light valve sound recording is of the variable density type, we shall consider only the latter.

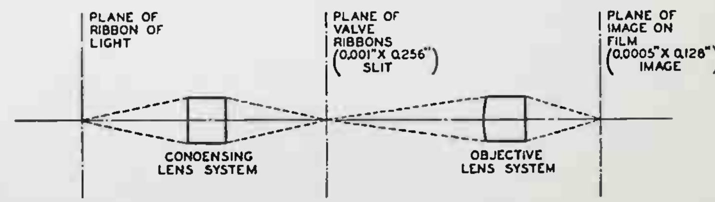

Fig. 5. Illustrating the specific changes in recording technic involved in the use of a 1-mil normal aperture (instead of 2 mils).

The remainder of this paper will be divided into a consideration of theoretical aspects of light valve modulation (Part II), a discussion of practical factors important in light valve design and use (Part III), and a general description of a new type of light valve (Part IV).

II. Theoretical Aspects of the Light Valve

1. VIBRATORY ACTION OF THE LIGHT VALVE

The light valve, considered in its electromechanical aspects, is similar in operation to the Einthoven string galvanometer. Considered in a transverse section (Fig. 6), a current I flows in a conductor A in a uniform magnetic field. (A second conductor B is used in the double-ribbon valve for the return of the current and moves always in opposite direction to the first, but the individual conductors may be considered to act separately.)

The conductor (ribbon) A will move at right angles to both the direction of the magnetic field and of the current, and the force F acting upon it is

$$F = kIH$$

where H is the intensity of the magnetic field linked by the conductor, and k is a constant. Thus the force acting upon the ribbon is proportional to (1) the strength of the magnetic field and (2) the instantaneous current. If I is an alternating current, such as corre-

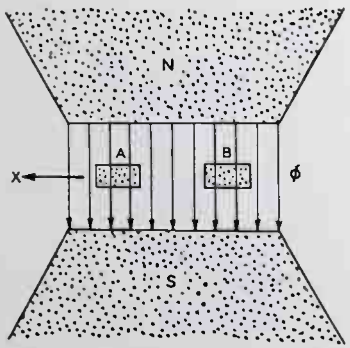

Fig. 6 Transverse section of light valve used in analyzing its action.

sponds to speech and music, the force F will be of similar character and the motion of the ribbon will alternate accordingly.

The extent of motion of the conductor per unit of current at low frequencies will depend solely on the total tensioning force exerted on the conductor as it lies stretched between its supports. That is, the force due to the current will deflect the ribbon until it is offset by an equal restoring force; the latter is a component of the tensioning force and is proportional to the displacement of the conductor from its line of support.

Under these conditions, the power input to the conductors is simply that dissipated by their resistance. The conductor, however, has uniformly distributed mass as well as elasticity, and the former becomes of increasing importance at high frequencies. The inertial

force of the moving conductor tends to keep the conductor moving and to offset the effect of the restoring force. As the influence of the mass increases, an increase in the motion of the conductor for a given amount of current occurs, and the valve becomes more sensitive or responsive. At a particular high frequency, the effect of distributed mass and elastance will offset each other and light valve "resonance" will occur. For this condition, the valve is highly sensitive; and for frequencies in the vicinity of resonance, distortion of the signal takes place in that the response is excessive compared with that at low frequencies. Fig. 7 shows relative response curves

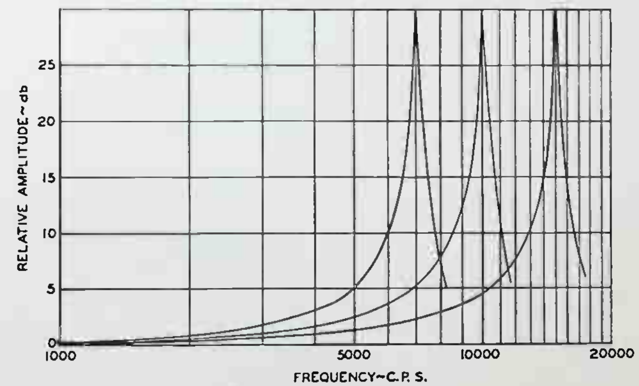

Fig. 7. Relative response curves of light valves whose resonances occur at 7000, 10,000, and 15,000 cycles.

of light valves whose resonances take place at 7000, 10,000, and 15,000 cycles, respectively.

2. LIGHT VALVE RESONANCE AND TUNING

This resonance, of course, is controllable and is generally caused to fall outside the useful range of recording. The resonance or "tuning" frequency is given[5] by

$$= \frac{1}{2l}\sqrt{\frac{T}{M}} = \frac{1}{2l}\sqrt{\frac{T}{A\rho}}$$

where
f = resonance frequency
l = length of vibrating conductor
T = tension
M = mass per unit length
A = area of conductor in cross-section
ρ = density of conductor material

The resonance or "tuning" frequency may be controlled, therefore, by changing the tension of the conductor, the length of its vibrating span, the cross-section of the conductor, or the density of its material. For a given design of light valve, l is fixed. For a given type of conductor ribbon, M, A, and ρ are fixed. Consequently, in practice the resonance frequency is set by adjusting the tension of the ribbon to a sufficiently high value.

3. IMPEDANCE OF THE LIGHT VALVE

The impedance characteristic of the light valve is of interest in considering the efficiency of the light valve. Since the motion of the conductors depends on the value of current flowing, the valve should be connected to its supply circuit under the most efficient

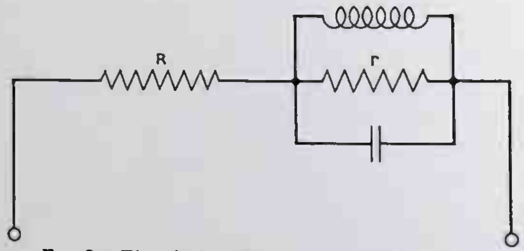

FIG. 8. Electrical equivalent of the light valve.

conditions, *i. e.*, when the impedance of the valve and the circuit are matched so that for a given emf. in the supply circuit, maximum current flows in the valve.

At low frequencies, the power delivered to the valve is entirely used in conductor heating, and yet the valve is most highly responsive when maximum current is delivered because the mechanical force set up is greatest. At high frequencies, the mass and elasticity of the conductor must be represented in the electrical impedance, for, as the conductors move, reaction emfs. are generated, which create reactive impedance in the valve circuit. At resonance, the reactances due to mass and elasticity offset each other and the impedance is controlled largely by the damping resistance (r), due to mechanical friction. Fig. 8 shows the electrical equivalent of a light valve. Fig. 9 shows a typical light valve frequency-impedance characteristic. The same data, when put in polar form (Fig. 10),

display the "motional impedance characteristic" typical of the telephone receiver and other vibrating instruments.

4. SCANNING LOSSES AND HARMONIC DISTORTION

The "Ribbon Velocity" Effect.—The foregoing discussion has dealt with the movement of the ribbons in response to an alternating current and, accordingly, with the variations of light flux passed through the valve. We are interested, however, in the variations

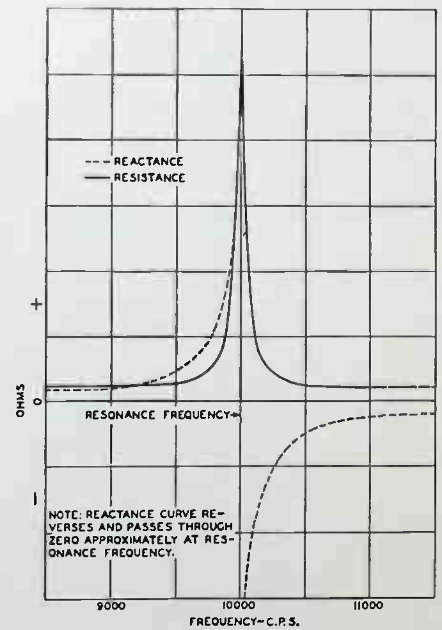

FIG. 9. The optical light valve frequency-impedance characteristic.

in point-to-point exposure of the moving film on which the light valve slit is imaged in a recording machine. The exposure given to the film, it may be readily seen, is not determined by changes in intensity of the light flux, but by the time required for any point on the film sound track to pass through the image of the light valve slit. This time, and the effective exposure of any point on the film, is therefore affected by the film velocity.

If the film moves very rapidly, the average exposure of the sound track will be low, and *vice versa*. The brilliancy of the lamp source, the condensing lens system, and the average opening of the light valve must be arranged, for any given film speed (*e. g.*, 90 ft. per min.), to give the proper average film exposure.

If the frequency being recorded is low, so that the velocity of the ribbons is small compared with the velocity of the film, the variations in film exposure will represent faithfully a pattern of the light

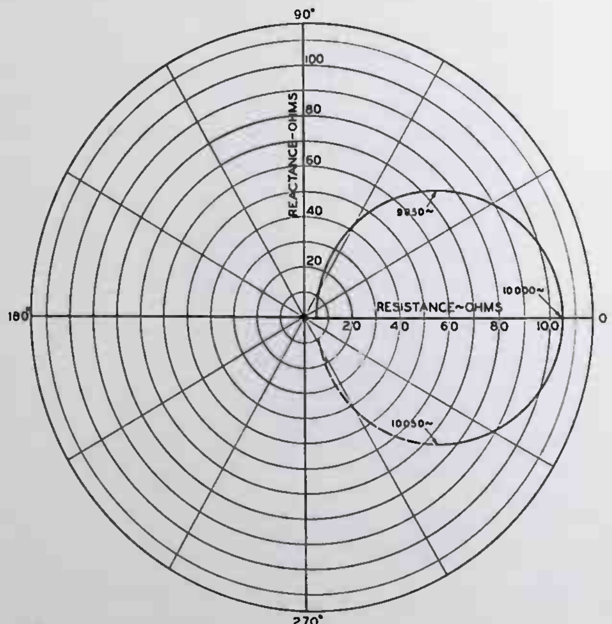

Fig. 10. Same data as in Fig. 9, plotted in polar coördinates.

valve modulation. As the frequency becomes high enough, however, the velocity of the ribbons increases, so that "the ribbon velocity effect," as it is called, comes into play. This results (1) in a loss of effective variation in exposure, which means a loss of recorded volume, and (2) in a degradation of wave-shape which includes the production of spurious harmonic frequencies.

The ribbon velocity effect is somewhat different in the cases of the single- and double-ribbon valves. It may be analyzed[6] as follows:

Double-Ribbon Case.—In Fig. 11, let a transverse line (infinitesimal striation) P of a film moving with velocity v be, at any time t, at the center of the exposure image, and let the instantaneous width of the image be $2w$. The half-width of the image is then w. Let the half-width of the image at that previous time t_1, when P just entered the image, be w_1, and let the half-width at that subsequent time t_2, when P will leave the image, be w_2. It will be assumed that the film velocity v always is greater than the rate of change of the half-image size (dw/dt).

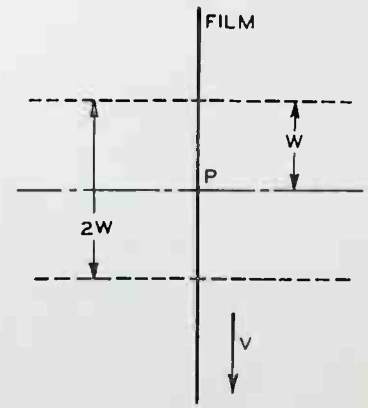

Fig. 11. "Ribbon velocity," effect diagram.

The total light received by P is proportional to

$$w_1 + w_2 = v(t_2 - t_1)$$

This must be expressed in terms of t.

Now

$$w_1 = v(t - t_1)$$

and

$$w_2 = v(t_2 - t)$$

If the image varies sinusoidally, that is,

$$w = a + b \sin \omega t$$

then

$$vt_1 = vt - a - b \sin \omega t_1$$

and

$$vt_2 = vt + a + b \sin \omega t_2$$

or, multiplying by ω/v for convenience

$$\omega t_1 = \omega(t - a/v) - (b\,\omega/v)\sin\omega t_1$$

and

$$\omega t_2 = \omega(t + a/v) + (b\,\omega/v)\sin\omega t_2$$

These equations are of the type

$$x = y + \alpha\sin x$$

so that x and y are odd functions of each other. Hence $x-y$ can be expanded into a Fourier series of y, containing only sine terms, *i. e.*,

$$x - y = \sum_{n=1}^{\infty} a_n \sin ny$$

Hence

$$A_n = \frac{2}{\pi}\int_0^{\pi}(x-y)\sin ny\,dy$$

Integrating by parts,

$$A_n = \frac{2}{n\pi}\left[-(x-y)\cos ny\Big|_0^{\pi} + \int_0^{\pi}\cos ny\,d(x-y)\right]$$

The integrated term vanishes since $x = y$ for both 0 and π; also

$$\int_0^{\pi}\cos ny\,dy = 0$$

Hence, putting

$$x - \alpha\sin x = y$$

$$A_n = \frac{2}{n\pi}\int_0^{\pi}\cos n(x - \alpha\sin x)\,dx$$

$$= \frac{2}{n}J_n(n\alpha)$$

by the Bessel integral.

Thus, we obtain the solutions

$$\omega t_1 = \omega(t - a/v) + 2\sum_{n=1}^{\infty}\frac{1}{n}J_n\left(-\frac{nb\omega}{v}\right)\sin n\omega\left(t - \frac{a}{v}\right)$$

and

$$\omega t_2 = \omega(t + a/v) + 2\sum_{n=1}^{\infty}\frac{1}{n}J_n\left(+\frac{nb\omega}{v}\right)\sin n\omega\left(t + \frac{a}{v}\right)$$

Whence

$$v(t_2 - t_1) = 2a + \frac{2v}{\omega}\frac{1}{n}\left[J_n\left(\frac{nb\omega}{v}\right)\sin n\omega\left(t + \frac{a}{v}\right) - J_n\left(-\frac{nb\omega}{v}\right)\sin n\omega\left(t - \frac{a}{v}\right)\right]$$

By expanding and regrouping,

$$v(t_2 - t_1) = 2a + \frac{4v}{\omega}\left[J_1\left(\frac{b\omega}{v}\right)\cos\frac{a\omega}{v}\sin\omega t + \frac{1}{2}J_2\left(\frac{2b\omega}{v}\right)\sin\frac{2a\omega}{v}\cos 2\omega t + \frac{1}{3}J_3\left(\frac{3b\omega}{v}\right)\cos\frac{3a\omega}{v}\sin 3\omega t + \ldots\ldots\right]$$

where $2a$ is the normal image width and $\frac{b}{a}$ the fractional modulation

Single-Ribbon Case.—It may readily be shown that the character of the alternating exposure is not affected by the direction of motion of the film relative to the fixed edge of the image. We shall assume that the film approaches the fixed edge of the image first. Whence, from the equation for ωt_2 above,

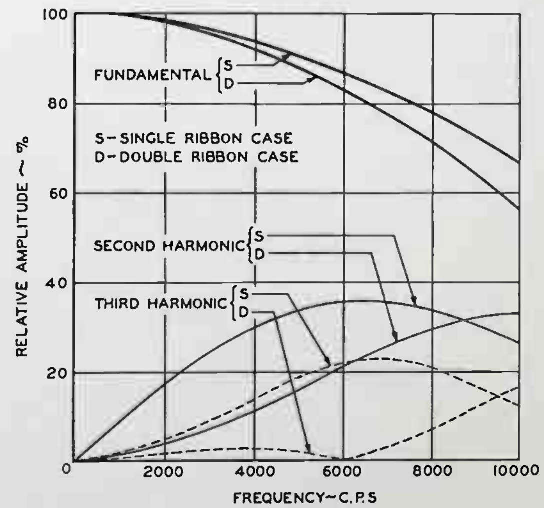

FIG. 12. "Ribbon velocity," effect for 0.5-mil normal image.

$$v(t_2 - t) = a + b\frac{2v}{b\omega}\left[J_1\left(\frac{b\omega}{v}\right)\sin\omega\left(t + \frac{a}{v}\right) + \frac{1}{2}J_2\left(\frac{2b\omega}{v}\right)\sin 2\omega\left(t + \frac{a}{v}\right) + \frac{1}{3}J_3\left(\frac{3b\omega}{v}\right)\sin 3\omega\left(t + \frac{a}{v}\right) + \ldots\ldots\right]$$

From the formulas, typical curves may be drawn which show the loss of amplitude of the fundamental component of the film exposure with increasing frequency, and the magnitudes of the various harmonics.

Such curves are shown for the case of a normal 0.5-mil image for the single- and double-ribbon valve, in Fig. 12. Here it will be noted that for 100 per cent modulation, the fundamental of both types of valve suffers a loss of several decibels at 10,000 cycles, the double-ribbon valve suffering about a decibel more than the single-ribbon valve.

In the matter of harmonic distortion, however, the double-ribbon valve is markedly superior, and this is especially true for the third harmonic, which relatively is very weak in the double-ribbon valve.

At lower modulations, the frequency characteristic of the fundamental improves, in the case of the single-ribbon valve, more rapidly

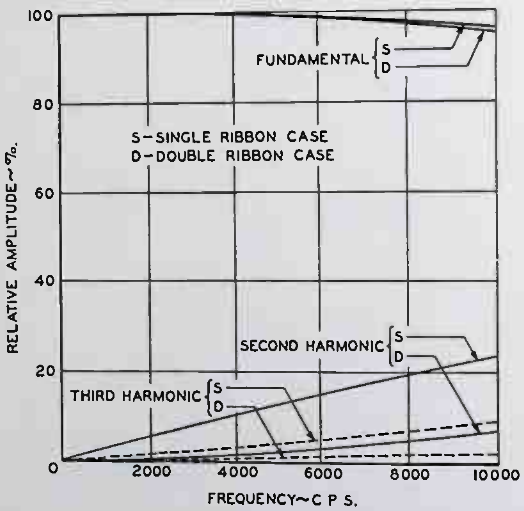

Fig. 13. "Ribbon velocity," effect for 0.167-mil normal image.

than for the double-ribbon valve, but the situation on harmonic distortion remains relatively much the same.

The diminution of the fundamental at high frequencies is of minor importance, because the influence of light valve resonance in present or future practice may be considered to offset it.

The illustration chosen (for 100 per cent modulation) is fairly typical of most recording situations where "noise reduction" apparatus is employed. It should be pointed out that, for the low valve spacings obtained on weak sounds with such apparatus, distortion of fundamental and harmonic production from the causes mentioned is greatly reduced. Fig. 13 shows curves corresponding

to those of Fig. 12, except that the valve spacing is reduced to 0.3 its former value.

This latter point is especially important in comparing the light valve with other light modulating devices, such as the flashing lamp, for it indicates that by reducing the amount of the average exposure, the light valve distortion may be reduced accordingly.

III. Practical Aspects of the Light Valve

1. FACTORS GOVERNING SENSITIVITY OF VALVE

For any given normal separation of the ribbons, the sensitivity of the light valve depends on (1) the force on the ribbons per unit current, and (2) the deflection of the ribbons per unit force.

The force on the ribbons per unit current depends, as we have seen, on the strength of the field in which the ribbons move. Aside from the use of (*a*) magnetic material having high permeability and (*b*) an efficient winding, in the case of an electromagnetic field; or the use of material having high residual permeability in the case of a permanent magnet field; the principal factor influencing the magnetic field is the length of the air gap. The air gap must be wide enough to accommodate the moving ribbon or ribbons and any additional light barrier placed between the magnetic poles.

In general, then, a valve of the one-plane type is more efficient magnetically than a valve of the two-plane type, for in the latter the air gap must, in general, be somewhat longer. Although in practice the magnetic yoke is brought to a high saturation point, so long as the reluctance of the air gap forms an appreciable part of the total reluctance, the magnetic efficiency of the circuit will be greater with a narrower gap. If this is put on the basis that a definite magnetic flux is required through the gap, then, with the narrower gap generally pertaining to the one-plane type of valve, the field magnetizing current required is smaller.

In considering the sensitivity of the valve for a given strength of magnetic field the following factors are important:

(*a*) *Low Resistivity of the Conductor Material.*—Since the deflecting force of the ribbons depends on the current flowing through them, the amount of a-c. power which must be supplied to the ribbons for a given deflection is obviously proportional to the resistivity of the ribbon material. For frequencies substantially below the resonance frequency, the impedance of the valve is closely equal to its d-c. resistance. The power required to drive the ribbons is therefore similar to that dissipated in the ribbon as a conductor. It is assumed in this discussion that the valve input transformer matches closely the valve impedance. Under this condition each doubling of the resistivity means a doubling of the power supplied to the valve per unit of current in the ribbons and therefore a loss of 3 decibels in sensitivity.

(*b*) *Resonance Frequency of the Valve.*—From the formulas, given in Part II, for the resonance frequency of a light valve, it is seen that the tension which must be applied to the ribbon is proportional to the square of the resonance frequency. This means that the higher the tuning frequency the less sensitive the valve, for the amount of the tension determines the size of the restoring force which tends to prevent displacement of the ribbons. With any given ribbon material, therefore, a doubling of the tuning frequency means a loss of 12 decibels in valve sensitivity.

(*c*) *Density of the Ribbon Material.*—The density, or specific gravity, of the ribbon material has an influence on the sensitivity of the valve. If two valves be alike except for the material of which their ribbons are composed, and if each be tuned to the same frequency, it is obvious, from the formulas, that the tensioning force will be greater for the valve having ribbon material of higher density. The tension required for any given resonance frequency will be proportional to the density of the material, and, therefore, the sensitivity of the valve varies inversely as the density of its conductor material. This means that each doubling of the density of the ribbon material causes a loss of 6 decibels in sensitivity.

(*d*) *Length of Vibrating Span.*—In considering the influence upon sensitivity of the length of span of the vibrating ribbon, it is necessary to consider only the resistance of the conductor material. If the length of span be doubled, the power supplied to the ribbon must be doubled; that is, there is a loss of 3 decibels in sensitivity. While it is true that the force created in the conductor by its reaction in the magnetic field is proportional to the length of the vibrating span, this increase in force is directly offset by the fact that the force doubled must move a conductor which, for any given tuning frequency, has a total restoring force proportional to the length of the vibrating span. That is to say, doubling the length of the span quadruples the tension for a given tuning frequency, but halves the

angular displacement of the ribbon. The net result, therefore, is that the sensitivity of the valve varies inversely as the square root of the length of the vibrating span.

(*e*) *Sensitivity of Single-Ribbon and Double-Ribbon Valves.*—The length of ribbon required in the double-ribbon valve is fundamentally twice that required in the single-ribbon valve. Therefore, for a given current in the vibrating ribbons, twice as much power must be supplied to the double-ribbon valve. This means an apparent loss of 3 decibels in sensitivity. However, the displacement obtained from two ribbons in the double-ribbon valve is, of course, twice that obtained with the same current in the single-ribbon valve. Therefore, for a given percentage modulation of the recording illumination, a factor of 6 decibels must be added in favor of the double-ribbon valve. The net result is that the double-ribbon valve is inherently 3 decibels more sensitive, for a given percentage of light modulation and consequent volume of reproduced sound, than the single-ribbon valve. This figure, of course, assumes valves which are alike in other design details, such as the nature of the conductor material employed, the flux density of the air gap, *etc.* This estimate of 3 decibels is conservative, for it assumes that ribbon material of the same cross-section is employed in either type of valve. Since the ribbon of the single-ribbon valve must be displaced twice as far as either of the ribbons of the double-ribbon valve, and since the width of the vibrating conductor is determined primarily by considerations of mechanical tolerances in relation to the amount of ribbon displacement required, it is more fundamentally correct to assume that in the single-ribbon light valve, for a fair comparison, the ribbon material should be twice as wide. If this assumption is made, the wider ribbon is equivalent to two of the narrower ribbons, vibrating side by side, and a further factor of 3 decibels should be allowed for the additional power required to displace the heavier ribbon. Thus, from a fundamental design standpoint the single-ribbon valve is 6 decibels lower in efficiency than the double-ribbon valve.

2. PROPERTIES OF LIGHT VALVE RIBBON

It is of major importance in the successful use of the light valve that the metal ribbon or tape used to form the vibrating light gate shall be adequate for the purpose it is to serve. In general, the ribbon should possess the following properties: (1) low resistivity,

(2) low specific gravity, (3) high tensile strength, (4) straightness of ribbon edges, (5) stability under continuous tension, (6) non-corrosiveness, and (7) non-magnetic character. The importance of (1), (2), and (3) have been discussed.

The importance of straight optical edges is apparent when it is considered that the variations from straightness cause changes from point to point in the light valve slit width and hence in average film exposure. In an average slit width of 1 mil an effort is made to keep edge straightness deviations below 0.1 mil. This represents a change of 10 per cent in average exposure for the corresponding portion of the sound track. This does not ordinarily mean a change in signal volume recorded, because the actual displacement of the ribbons is unaltered; but it means a slight shifting, from point to point along the light valve, of the exposure in relation to the straight-line part of the H & D curve. It can also affect the maximum recordable volume by altering the clash point of the valve.

Among the materials which are suitable for use duralumin has been found greatly superior and has, in addition, proved to be fairly workable material. The following table shows the more important constants of various metals which might be considered.

Constants of Various Metals Used for Light Valve Ribbons

Material	Tensile Strength	Density	Resistivity	Tensile Strength Density	Figures of Merit: Sensitivity	Figures of Merit: Breaking Frequency
Aluminum	27,600	2.7	3.0	10,200	1.29	0.48
Aluminum (90% Cu) Bronze (10% Al)	90,000*	8.3	2.0	10,800	0.51	0.51
Copper (hard drawn)	65,500	8.93	1.8	7,350	0.50	0.35
Duralumin	75,000*	2.8	4.6	26,800	1.00	1.27*
Duralumin (light valve ribbon)	59,000	2.8	4.6	21,000	1.00	1.00
Molybdenum	154,000†	10.0	5.7	15,400	0.25	0.73
Molybdenum (0.002 wire)	200,000*†	9.0	5.7	22,000	0.28	1.05*
Silver	42,600	10.5	1.6	4,050	0.50	0.19
Tungsten	590,000*†	18.8	5.5	31,400	0.14	1.50*
Tungsten (ribbon)	450,000†	18.8	5.5	24,000	0.14	1.14

* Probably less for ribbon form.
† Material difficult to work smoothly.

The figure of merit for sensitivity indicates directly the comparative sensitivity of valves employing the different materials, and is

obtained by multiplying the ratio of the densities by the square root of the ratio of the resistivities. The figure of merit for breaking frequency indicates directly the ratio of maximum allowable tuning frequencies for light valves using the various metals. Where the figures given are not otherwise noted, they are for the metal in bar form, and it should be realized that neither the tensile strength of metal in this form nor that in drawn wire form may, as a general thing, be realized in the case of metallic ribbon of the dimensions required for light valves. It is readily seen that duralumin is the only metal listed which has a high figure of merit for both sensitivity and breaking frequency. To give specific illustrations, a light valve of any character employing molybdenum must be inherently about 12 decibels less sensitive than one employing duralumin, and a light valve employing tungsten would have a loss of sensitivity, in

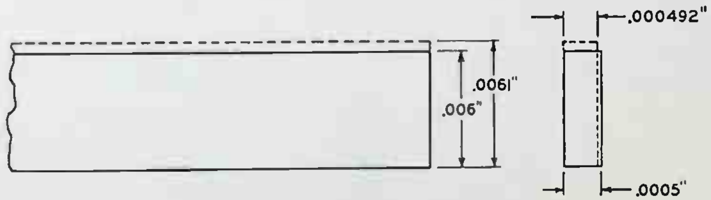

FIG. 14. Cross-section diagram of light valve ribbon.

comparison with one using duralumin, of 17 decibels. The composition of duralumin varies somewhat; the alloy at present used for light valves has the following composition:

Aluminum	94
Copper	4
Manganese	0.5
Magnesium	0.5
Silicon, iron, *etc.*	1

As is well known, heat treatment and aging have an influence on the tensile properties of duralumin, and much effort has been expended in recent years to increase the tensile strength. As a result of such efforts, the ribbon now employed has a tensile strength about 75 per cent greater than that of the earlier light valve ribbon.

There are two general methods which may be employed for the production of duralumin ribbon. In the first, wire is drawn to the proper cross-section and flattened into the ribbon form by rolling;

in the second, ribbons 0.006 inch wide are sheared directly from sheets of duralumin foil 0.0005 inch thick.

In the rolling method, the primary obstacles are the extreme accuracy required of the rolls and the uniformity throughout its length of the material. Fig. 14 represents a cross-section of ribbon 6 mils by 0.5 mil. If manufacturing tolerances hold the width to

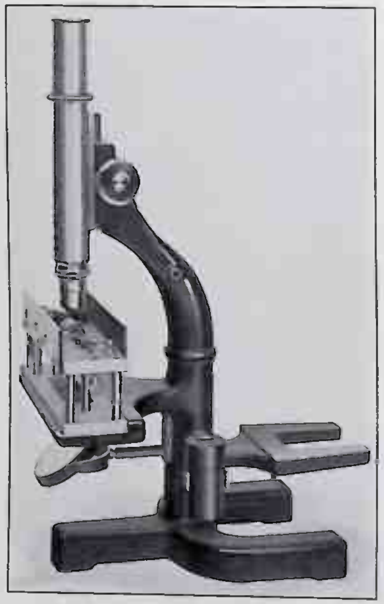

FIG. 15. Illustrating a microscope fixture attached for visual examination of ribbon edges.

± 0.1 mil and the density of the material remains constant, the thickness must be held to 0.008 mil, or eight one-millionths of an inch. In the shearing process accurate alignment of cutting shears must be supplemented by a technic for producing foil of uniform thickness, free from pinholes, embedded impurities, *etc*. Because of the empirical nature of alloy processing, it is customary, as a check inspection after regular manufacturing inspection has been completed, to test a substantial portion, about 5 per cent, of all supposedly satisfactory ribbon. Valves are actually strung with this material

which is then inspected for ribbon edge straightness, and the ribbons are tuned to destruction to determine their breaking point. During the past two and one-half years approximately 2500 valves have been thus strung and inspected. Fig. 15 shows a microscope fixture attached for visual examination of ribbon edges.

Fig. 16 shows an illustration of rough edges, taken from an early grade of ribbon. Fig. 17 shows a type of variation in ribbon due to undulating edges. Fig. 18 shows a sample of ribbon excellent in its quality of edge straightness.

3. LIGHT VALVE "HYSTERESIS" AND RIBBON SLIPPAGE

In much of the earlier studio recording, a phenomenon was often noticed called light valve "hysteresis." This was due to the failure

FIG. 16. Light valve ribbon having rough edges.

FIG. 17. Light valve ribbon having wavy edges.

FIG. 18. Light valve ribbon having straight edges.

of the valve ribbons to return, after a large impulsive displacement, to their original normal position. This gave the d-c. amplitude characteristic, for wide variations in current, the appearance of a magnetic hysteresis loop. Fig. 19 shows the hysteresis loops for an older type of valve for two tunings, at 7000 and 10,000 cycles, respectively. From these curves, it is seen that higher tuning reduces the magnitude of the hysteresis effect, if we judge this magnitude by the displacement (in mils) of the opposite sides of the loop. This might be expected, because the vertical component of the ribbon tension, tending to hold the ribbon in place by means of friction at the ribbon supports, is increased in proportion to the increased tension required for the higher tuning frequencies.

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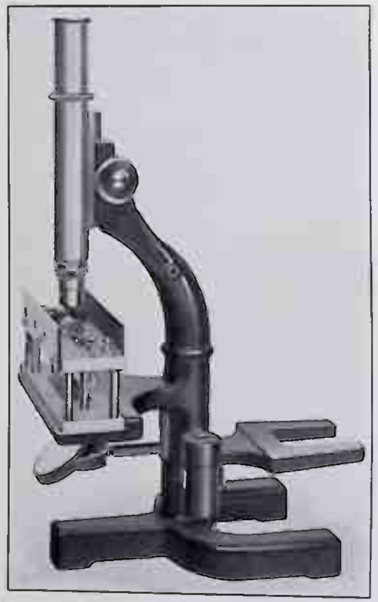

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Improvements in recording technic required that the valve tuning frequency be raised considerably above the earlier value of 7000 cycles. This development, in combination with the availability of the recently developed stronger type of duralumin ribbon, permitted the general adoption of higher tuning frequencies, which automatically reduced the hysteresis effect. However, with the advent of noise reduction equipment to reduce film background

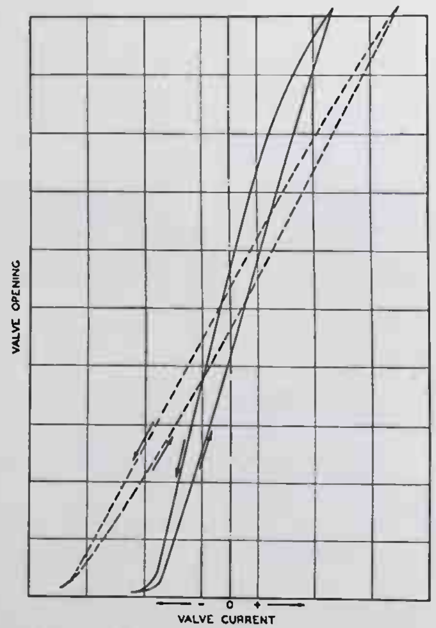

FIG. 19. Light valve hysteresis (old valve).

noise, the requirements for the exact biasing of the valve ribbons to small average slit widths made necessary the elimination of most of the hysteresis tolerated in the older equipment.

Various methods were tried experimentally to secure greater stability of average valve spacing, such as pin locating stops, metal clamps, paper spacers, cement, *etc.* The cement method was given a field trial, but did not prove satisfactory for general use.

The valve modification adopted for general field use practically eliminated the hysteresis effect and was simple enough to enable

the ready conversion of available light valves; the slight modification of the bridge support and spacing pincers, shown in Fig. 20, has practically eliminated the hysteresis effect. The curves of Fig. 21 demonstrate this for 7500 and 10,500 cycles' tuning.

In interpreting the true importance of light valve "hysteresis," we must realize that it is only superficially like magnetic hysteresis. In the first place, it does not represent a loss in energy. Secondly, it is generally not present except for current cycles which exceed a critical (and large) value; thus, for the smaller displacements, frictional anchoring forces at the ribbon supports are adequate. Thirdly, if the sides of the hysteresis loop are straight and parallel, the spurious harmonics produced are small.

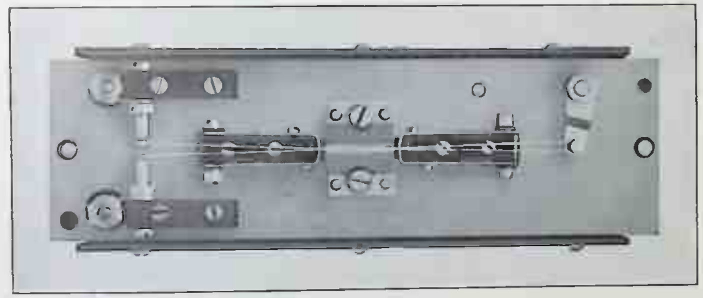

FIG. 20. Illustrating the slight modification of the bridge support and spacing pincers for eliminating hysteresis.

The principal detrimental effects of ribbon slippage were therefore (1) material variation in average exposure of the film, which interfered with exact sensitometric control, and (2) departures from normal of the load capacity of the valve, which interfered with standardization of recording technic and the securing of maximum recorded volume range. It is for both of these reasons that the introduction of noise reduction equipment required the reduction of light valve hysteresis.

4. AZIMUTH AND FOCUSING ERRORS IN RECORDING

The light valve is an electromagnetic shutter, and it translates the amplified electrical energy from the microphone into mechanical energy in such a manner that the light passing through the valve is proportional to the speech waves impinging upon the diaphragm of the microphone. It is, therefore, necessary to photograph the light valve action as accurately as possible. This is made possible by two important adjustments, the first of which is the azimuth adjustment of the light valve, and the second, the focal adjustment of the objective lens.

The azimuth adjustment of the light valve consists in locating the horizontal plane of the valve perpendicular to the direction in which the film is traveling. This adjustment also positions the

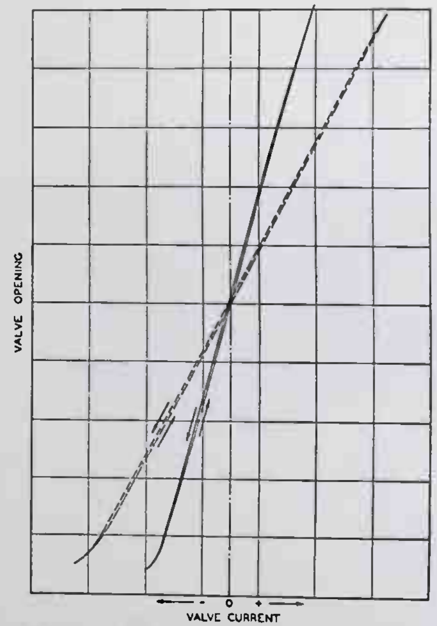

FIG. 21. Light valve hysteresis (improved valve).

striations on the film so that they are perpendicular to the direction of the film travel. An error in the azimuth adjustment of the light valve produces an azimuth deviation on the recorded film.

The azimuth deviation on the recorded film must be considered in relation to the azimuth deviation of the scanning image in the reproducer. Unless these values of azimuth deviation are identical in degree and direction, the losses at the higher frequencies are greater than those for optimum conditions of adjustment. If,

however, the azimuth deviation of the film varies from that present in the reproducer, we have to deal with the sums and differences in the deviation of each to obtain the effective value.

Experimental measurements of azimuth deviations in both the recorded film and the scanning image in the reproducer have indicated that the effect of an azimuth deviation on the recorded film and no azimuth deviation in the scanning image is equivalent to a similar deviation of the scanning image and no deviation in the recorded film for small values of azimuth deviation. The effect of the azimuth deviation of the scanning image has been treated both theoretically and experimentally and presented in a paper[7] before this Society.

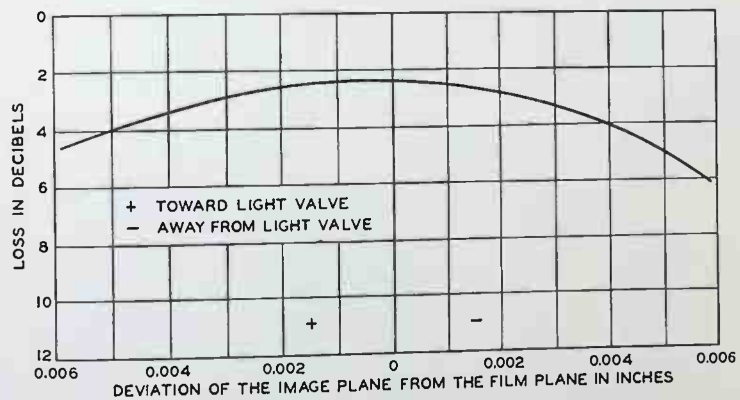

FIG. 22. Influence of improper focus on reading, *etc.*

Figs. 6 and 8 in that paper show the loss for various azimuth deviations of the 0.0005-inch light valve image.

The adjustment of the $f/1.5$ objective lens consists in the movement of the lens along the optical axis until the light valve ribbons are focused on the film emulsion at a reduction of 2:1.

As the objective lens system is moved along the optical axis, the plane of the image of the light valve ribbons also moves, but at a slower rate than that of the objective lens. Fig. 22 illustrates the influence of improper focus in the recording of a 7000-cycle sound track. A 2- and a 4-mil deviation of the image plane from the film plane results in an approximate additional loss of 1 and 3 decibels, respectively, at 7000 cycles. A more general expression of the effect of the microphone. It is, therefore, necessary to photograph the light valve action as accurately as possible. This is made possible by two important adjustments, the first of which is the azimuth adjustment of the light valve, and the second, the focal adjustment of the objective lens.

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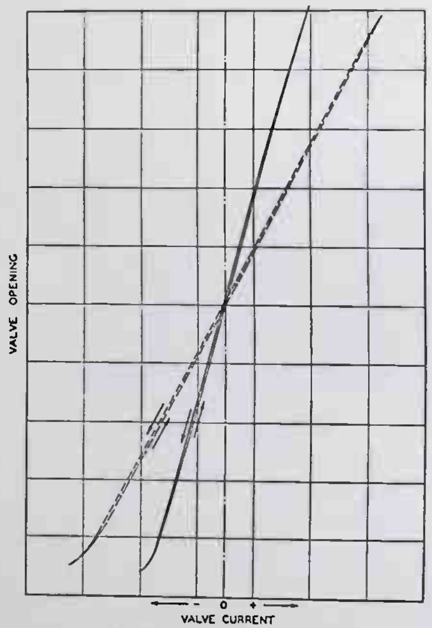

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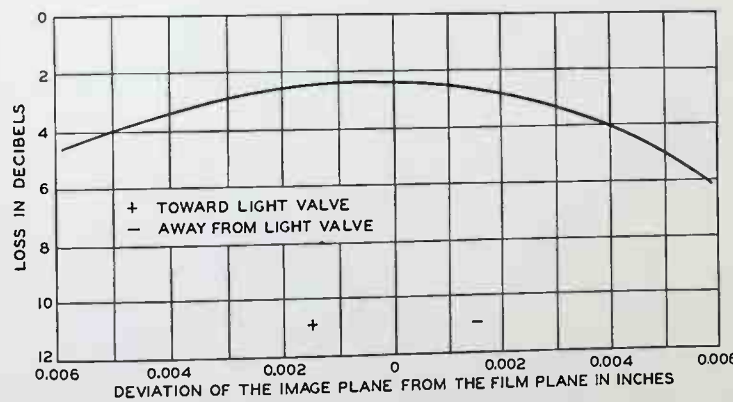

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As shown in the paper by Stryker,[7] when the loss due to both improper focus or an increase in the average image width and the azimuth deviation occur simultaneously, as they may in practice, the total loss of reproduction due to the two of them jointly will be the sum of the individual losses produced by each separately.

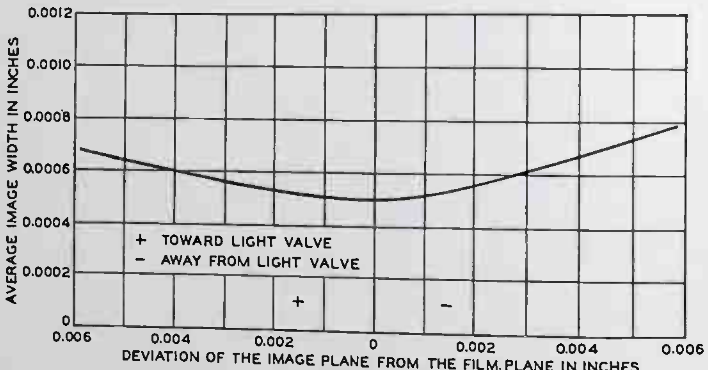

Fig. 23. Effect of improper focus on average image width.

It is, therefore, apparent that either the azimuth adjustment of the light valve or the focal adjustment of the objective lens, or both, are important factors that may seriously affect the quality of reproduced sound records. If the analysis of light valve action given in Part II is to hold, the average image width assumed must be consistent with the azimuth and focus conditions.

5. TUNING METHODS

An amplitude resonance curve for a light valve is shown by Fig. 7. The resonance peak indicates that the valve requires 30 decibels less power at the resonant frequency than at the low frequencies for the same modulation.

Several methods of measuring the resonant frequency of light valves have been employed. A visual method, used to tune the

earlier telephoto and sound film valves utilized a microscope to observe the maximum deflection of the ribbons when the frequency of an oscillator connected to the ribbons was varied.

The visual method proved satisfactory until widespread use of recording equipment placed exacting limits on the resonant fre-

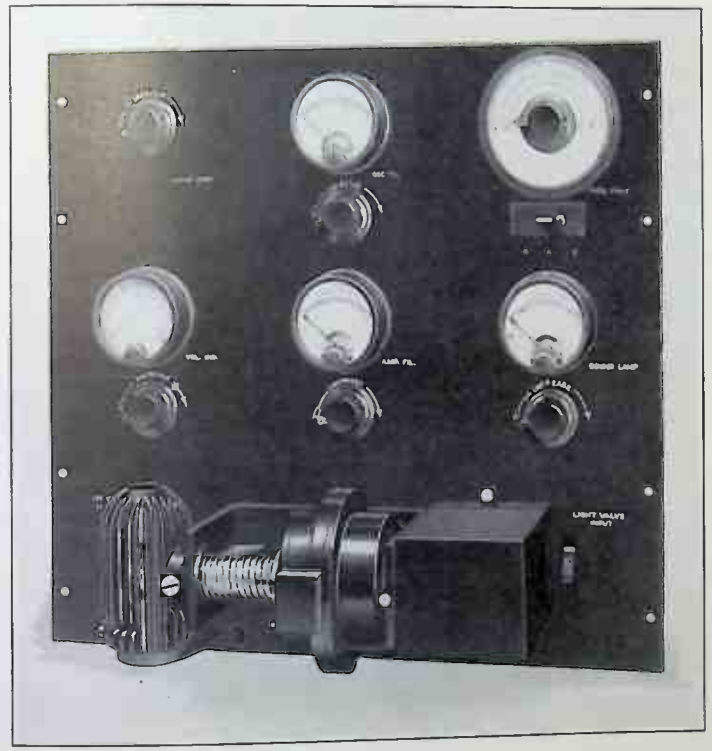

Fig. 24. Separate valve tuning unit supplied with Western Electric recording equipment.

quency, especially where it became desirable to have several valves operate at nearly the same clash point. The sound recorder itself provided ready means to determine the tuning frequency more accurately than the visual method. The procedure was to use the photoelectric cell monitoring system to measure the degree of light modulation by the valve when known current levels were supplied to the valve from a variable oscillator. An output level variation of the monitoring circuit of ±0.5 decibel could be measured with the ordinary volume indicator method and, therefore, from the resonance curve (Fig. 7) it is seen that it is possible to measure the tuning frequency to within ±50 cycles from the resonant frequency.

The use of recording equipment for tuning more than a few valves would have proved inefficient; and, therefore, separate valve tuning units were supplied with Western Electric recording equipment. This equipment is shown in Fig. 24. An oscillator, amplifier rectifier, light source, light valve field coil, and photoelectric cell are mounted on a 17- by 19-inch panel. The schematic circuit of Fig. 25 shows the

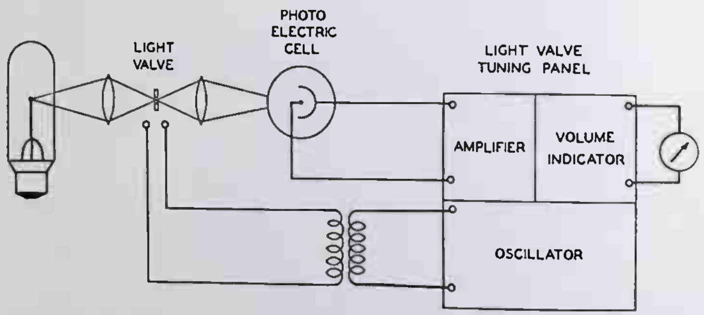

Fig. 25. A schematic circuit of the tuning unit of Fig. 24, showing the valve operation in the tuning circuit to be similar to that in an actual recording circuit.

valve operation in the tuning circuit to be similar to that in the actual recording circuit.

Another method of tuning valves, apparently due to R. D. Gibson, makes use of the motional impedance characteristic of the valve, shown in Fig. 10. The electrical impedance of the light valve with the magnetic field applied is equivalent to an anti-resonant electrical circuit as shown by Fig. 8. Many electrical methods of observing the tuning point of such circuits are familiar to the electrical art and a simple device consisting of an oscillator and a thermocouple in series with the valve has been described by Ceccarini.[8] Other arrangements employ volume indicators, thermocouples, or rectifiers to measure the voltage across the valve terminals when the valve is connected to an oscillator source supplying approximately constant current to the valve. It is evident that the series method of observ-

ing the peak will not indicate the correct peak sharpness unless the thermocouple resistance is small compared with the d-c. resistance of the light valve. Similarly, the voltage method will not indicate the correct sharpness unless the voltmeter impedance is high compared with the resonant impedance of the light valve.

6. LIGHT VALVE OVERLOAD AND CLASH

A phenomenon, concerning which little experimental evidence has been presented, is that of wave-form distortion due to light valve overload. In the two-plane type of valve the action which

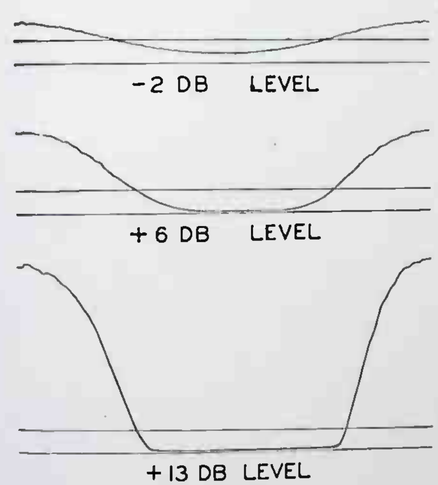

Fig. 26. Overload wave forms of single-plane valve for 100 cycles.

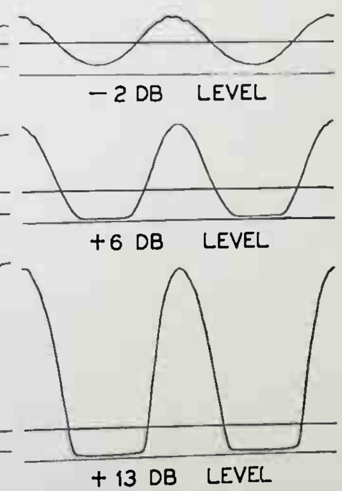

Fig. 27. Overload wave forms of single-plane valve for 250 cycles.

might be expected would be a simple cutting off of the negative troughs of the wave whenever the ribbons were sufficiently displaced so as to cut off all light. This assumes that, in the main, modification of the overload distortion, due to photographic considerations and due to non-linearity of valve displacement for excessive movements of the ribbon, might be neglected.

In the one-plane type of valve, however, it has not been clear what phenomena took place under similar conditions. Figs. 26, 27, and 28 are illuminating in this respect, as they show the relative wave-forms at frequencies of 100, 250, and 1000 cycles, respectively, which are encountered under the following conditions:

(*a*) modulation 2 db. below overload
(*b*) modulation 6 db. above overload
(*c*) modulation 13 db. above overload

It is readily seen in these cases that the type of distortion obtained is a relatively simple one, and is very much of the type that might be expected. Figs. 29, 30, and 31 show experimental wave-forms

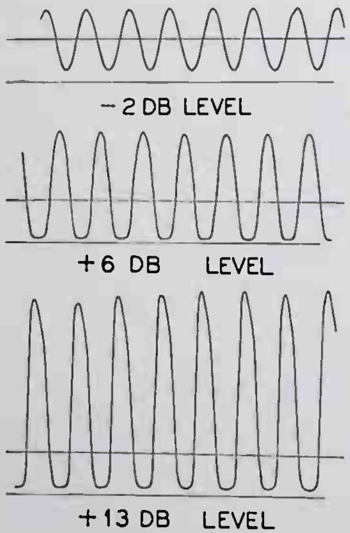

FIG. 28. Overload wave-forms for single-plane valve at 1000 cycles.

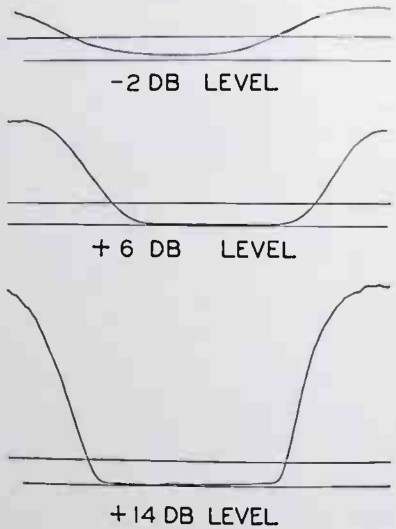

FIG. 29. Overload wave-forms of double-plane valve for 100 cycles.

obtained from a two-plane type of valve for the same frequencies, respectively, at the following levels:

(*a*) 2 db. below overload
(*b*) 6 db. above overload
(*c*) 14 db. above overload

It is quite apparent, however, from the overload wave-forms of both types of valve that distortion of this kind is highly objectionable from a sound quality standpoint, and is to be avoided in either type of valve by observing proper recording margins against overload.

Under average recording circumstances the frequency of occurrence of overload on speech and music sounds may be expected to follow the curve given by Sivian[9] for the relative distribution of instantaneous amplitudes of speech and music throughout the frequency spectrum. Thus, at very high frequencies sounds of high amplitude are seldom to be expected.

The relation of light valve overload to conductor heating and valve sensitivity should also be considered. If, for example, we

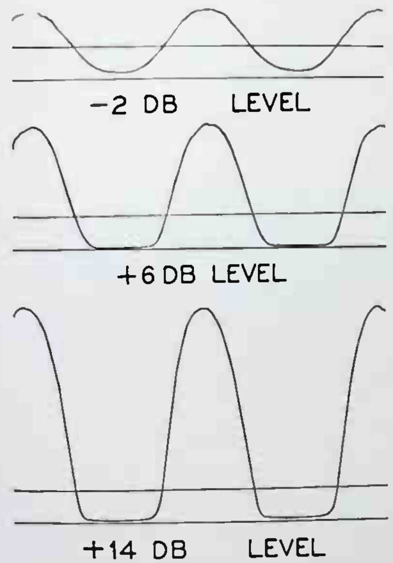

FIG. 30. Overload wave-forms of double-plane valve for 250 cycles.

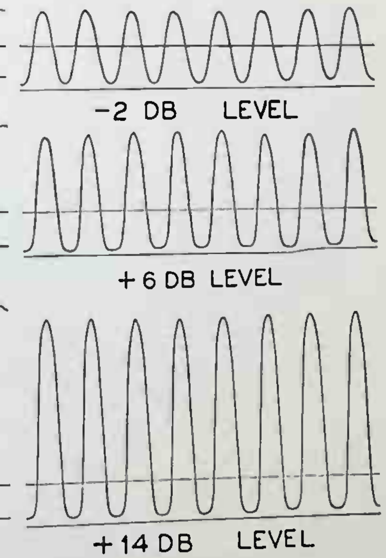

FIG. 31. Overload wave-forms of double-plane valve for 1000 cycles.

compare two types of valve differing in sensitivity by 15 decibels, it is obvious that the normal power required to bring the more efficient valve to the clash point is 15 decibels less than for the second valve. This means, of course, a much smaller internal heating of the valve since less power is dissipated. Consequently, one may expect a smaller temperature rise in relatively sensitive valves and correspondingly somewhat greater mechanical stability. One may also expect that the maximum overload level to which a valve may be subjected, from a temperature standpoint, relative to that level at which overload just takes place, will be largely dependent on the relative sensitivity of the valve.

IV. A Permanent Magnetic Type of Light Valve

A new type of light valve will now be described which represents several fundamental advances in light valve design. It is a valve which uses a minimum total ribbon length in operation so that its electrical efficiency is high. It is readily tuned to very high frequencies. It is, within experimental error, entirely free from hysteresis and ribbon slippage. It contains temperature compensation features to maintain greater constancy of spacing and tuning and, in general, is very rugged and stable. These ribbons are clamped in place in such a manner as to add to the constancy of spacing and

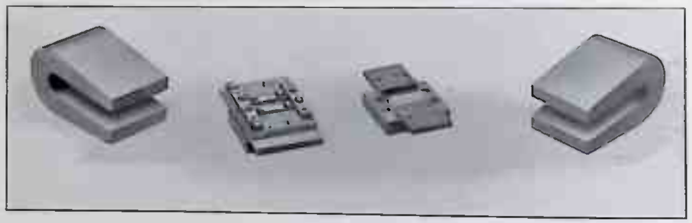

FIG. 32. The principal parts of a new type of light valve.

tuning. Possessing a permanent magnet field, it requires no field exciting current, yet a higher flux density is secured in its air gap than was accomplished electromagnetically in the earlier light valve. Last but not least, it is especially light and compact.

Fig. 32 shows the principal parts of this valve. On the right and on the left are two permanent magnets which fasten to the central portion of the valve, the objective and condenser sides of which unit are shown in the center of the figure. The size of the units of the valve may be estimated by considering that each of the magnets is $1^1/_2$ inches in length and that the base, as shown, of the top pole-piece is $^7/_8$ by $^3/_4$ inch. The ribbon when in place lies under the glyptol clamps on the condenser side of the valve. When the condenser side and the objective side of the valve are placed together the slit in the center of the latter lies opposite the aperture between the ribbons.

The valve in an assembled form with associated condensing lens is shown in Fig. 33. The total length of the unit as assembled there is $3^5/_8$ inches. Fig. 33 also shows in place the terminal strip to which connections are made from the string unit of the valve.

The light valve weighs 300 grams, or about 11 ounces. The permanent magnets account for about 220 grams of this weight, so that the central unit of the light valve comprising the two components shown in the center represents a weight of 80 grams or about 2.8 ounces.

The total amount of ribbon contained in the electrical circuit of the valve as used in recording is approximately 1 inch. For the

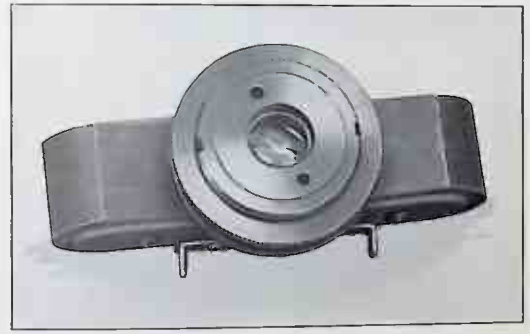

FIG. 33. The new valve in its assembled form with the associated condenser lens.

present studio light valve the corresponding length is about 8 inches. The valve can without difficulty be tuned to frequencies of the order of 12,500 cycles. Of course, as in any valve, tuning to such a high frequency can be accomplished only by a corresponding sacrifice in sensitivity. But the electrical and magnetic efficiency of the valve permit the use of relatively high tuning with a corresponding sensitivity equal to or greater than the present studio light valve when a considerably lower tuning frequency is used.

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PRINCIPLES OF MEASUREMENTS OF ROOM ACOUSTICS

BY

E. C. WENTE
Bell Telephone Laboratories

A METHOD OF DETERMINING ACOUSTIC PROPERTIES FROM A MEASUREMENT OF THE ACOUSTIC TRANSMISSION VERSUS FREQUENCY CHARACTERISTICS BETWEEN POINTS WITHIN THE ROOM

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PRINCIPLES OF MEASUREMENTS OF ROOM ACOUSTICS*

E. C. WENTE**

Summary.—*The acoustic characteristics of a room can in great part be evaluated from a knowledge of the rate with which sound in the room dies down when emission from the source ceases. The physical principles underlying the relationship are briefly discussed. It is shown by specific examples that we can obtain valuable additional information about acoustics of a room by recording the sound level at one or more points in the room when the frequency of the sound is continuously varied.*

The function of a motion picture sound system is to transmit an acoustic facsimile of sound generated in a studio to an audience in a theater. This transmission occurs over a series of acoustical, mechanical, optical, and electrical paths, distortion along any one of which will impair the quality of the received sound. The distortion along any part of the route, except those from the source to the microphone and from the loud speaker to the listener, can be determined from a measurement of the transmission efficiency at a number of discrete frequencies distributed throughout the audio-frequency range. Transmission along the acoustic paths is of such a totally different character that here a measurement of such type is of no practical value. For the proper adjustment of the acoustical paths reliance has been placed principally upon aural judgments. We can tell something about the character of sound transmission within a room if we know its reverberation time,† or, what is equivalent, the rate of decay of the transient tone when a steady tone is interrupted at the source. But whatever the reverberation time may be, the quality and level of the sound at different parts of a room may vary between quite wide limits. The determination of the acoustics of a room by measurement of the reverberation time is analogous to the determination of the characteristics of an electrical transmission line by

* Presented at the Spring, 1935, Meeting at Hollywood, Calif.; re-presented at the Fall, 1935, Meeting at Washington, D. C.

** Bell Telephone Laboratories, Inc., New York, N. Y.

† This is the time required for the average sound energy initially in a steady state to decrease to one-millionth of its initial value.

measurement of the transient current at the receiving end when a voltage is interrupted at the sending end, a method sometimes used before convenient electrical audio-frequency oscillators became available, but which is not nearly so satisfactory as a measurement of the transmission *vs.* frequency characteristic.

Transmission of signals over an ordinary electrical line differs from the transmission of sound between points in a room in two important respects: the transients are of much longer duration and, because of the fact that the air in a room is capable of oscillating at an exceedingly large number of different resonant modes within the audio-frequency range, the transient oscillations have practically the same frequency as the steady-state tone. It is for these reasons that measurements of transient oscillations have, on the whole, been of more practical value in the study of the acoustics of rooms than in the study

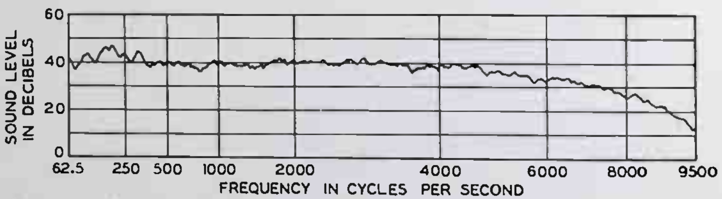

FIG. 1. Recorded frequency characteristic of sound transmission in a room when the frequency is varied rapidly.

of the characteristics of electrical systems. Since data on transient oscillations have only a limited value even in the case of sound in rooms, while a measurement of the transmission *vs.* frequency characteristic has been found to be a convenient and accurate method of evaluating the practical performance of an electrical system, it is worth while to investigate the possibilities of determining the character of sound transmission between points in a room by a more direct method.

A few measurements will convince any one that, in general, no useful information can be obtained from measurement of the transmission at a number of discrete frequencies, for the values will be found to vary by many decibels with only a slight shift in the frequency. More practical results are obtained if the instantaneous pressure levels are recorded while the frequency of the source is varied continuously. The high-speed level recorder, previously described, is particularly suitable for making such records. The method of

procedure in these measurements is to generate at one point in a room a tone of constant strength, but varying continuously in frequency, and record at a second point the sound pressure level by means of a microphone and the level recorder. When the recorder is set to operate at low speed, the readings represent the levels of the power averaged over a relatively long time-interval. If, then, the frequency is changed rather rapidly during the recording, the readings will represent levels of power averaged over a relatively wide frequency-

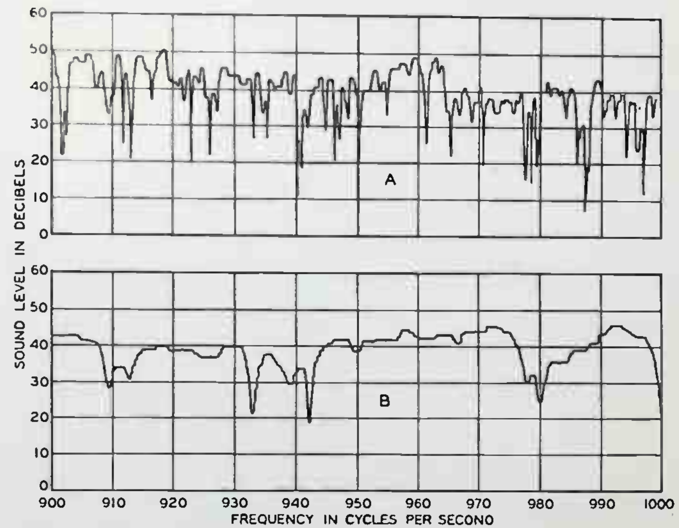

FIG. 2. Recorded frequency characteristic of sound transmission in a room when the frequency is varied slowly: (*A*) live room; (*B*) damped room.

interval. Fig. 1 shows a transmission curve obtained between two points in a room under these conditions. The frequency was varied at a rate such as to cover the whole designated range in about $1^1/_2$ minutes. A transmission characteristic of this type will show whether speech or music transmitted between the two points will have the proper balance between the high- and the low-frequency components. If the high frequencies predominate, the sound will be characterized by shrillness; and if the low frequencies overbalance, the sound will have a muffled quality. Other quality characteristics are indicated when there is a rise or a depression in the transmission curve at intermediate frequencies. Similarly, measurements in the studio can show something about the character of sound reproduction that would result from various placements of the pick-up microphone.

The transmission curve obtained in the manner just described does not tell a sufficiently complete story to enable us to say whether the transmission between the sending and receiving points for speech or

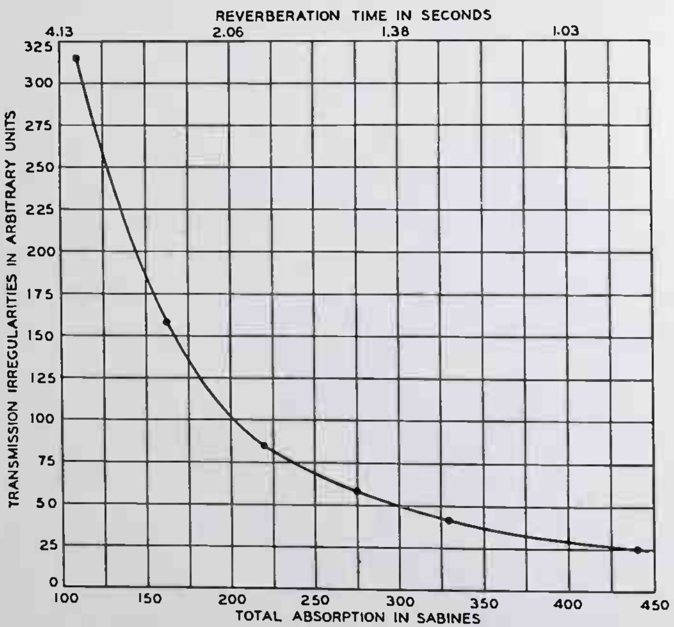

FIG. 3. Relation between irregularity of transmission *vs.* frequency characteristic and reverberation time; capacity of room, 10,000 cubic feet.

music will be the best possible, or even satisfactory. As far as we can tell from these curves, the room might be entirely too dead acoustically for music, or so live that received speech would be quite unintelligible. Nevertheless, if records taken in this manner show an improper balance, we are safe in concluding that transmission over the measured path will not be satisfactory.

Now let the operating speed of the recorder be set to a high value and the frequency be varied slowly while the positions of the microphone and the loud speaker and other conditions are kept unchanged.

Under these conditions the recorder will indicate levels of sound pressure averaged over a small frequency-interval. A portion of a curve so obtained is shown in the upper part of Fig. 2. The frequency range here given is only 100 cycles, extending from 900 to 1000 cps. Although the total change in frequency is only 10 per cent, we find that the curve has innumerable peaks and valleys and that the transmission varies through a range of at least 40 decibels. If, upon measurement, we should find any part of an electrical communication channel

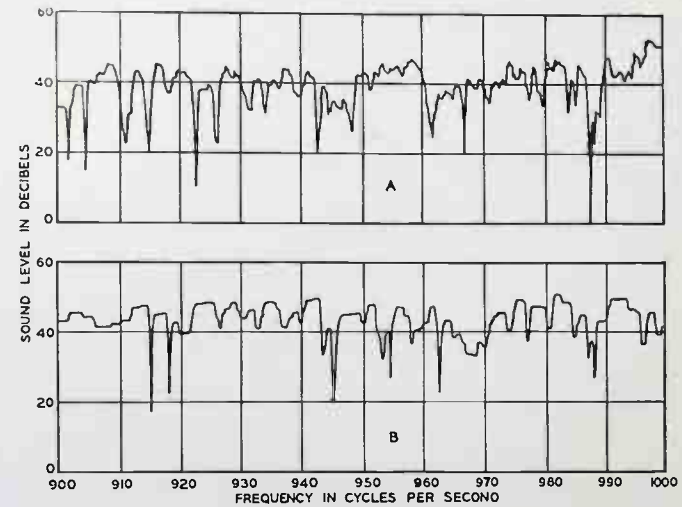

FIG. 4. Transmission *vs.* frequency characteristics to live and to dead parts of a room.

to have a characteristic as irregular as this one, we should probably conclude that the system would be incapable of transmitting speech intelligibly. As a matter of fact we should be correct in our conclusion, for this curve was taken in a room that was acoustically so reverberant that it was almost impossible to carry on a conversation between the sending and the receiving points.

Following these measurements the room was given an acoustic treatment by the introduction of sound-absorbing materials such that the transmission as judged by aural observation was practically ideal. All other conditions were kept the same. The transmission characteristic now obtained over the same frequency range is shown in the lower part of Fig. 2. It will be noted that there are relatively few large dips and that the number and extent of the small irregularities are greatly reduced. The change in the reverberant quality of the room is thus seen to be easily observable from the change in the character of these transmission curves. This change in character can be measured and related to the reverberant quality of the room, or, more accurately, to the reverberant quality of the sound transmission between any two points within the room. One method of evaluating the degree of irregularity of the curve in a given frequency-interval is to take the sum of the pressures of all the minimum points and subtract the result from the sum of the pressures of all the maximum points. In Fig. 3, values so obtained for a 100-cycle band-width are plotted against the reverberation time of the room. The observed points are seen to lie well along a smooth curve.

We might well ask, if there is such a close correlation between reverberation time and the degree of irregularity in the transmission curves, why not simply measure the reverberation time, which can be accomplished within a few minutes by means of the high-speed level recorder. The values plotted in Fig. 3 were obtained in a room in which the sound was fairly uniformly distributed, and both loud speaker and microphone were as far removed as possible from the neighborhood of absorbing surfaces. The reverberation time when measured at various points in a room will have about the same value, for it is determined primarily by the rate of decay of the sound density averaged throughout the whole room; but the degree of irregularity of the transmission to different points in the room can vary markedly, as it depends upon the configuration of the room and the distribution of the sound-absorbing surfaces. The difference is illustrated by the curves of Fig. 4, which were obtained with two different microphone placements in the same room. For the upper curve the microphone was located in a part of the room where most of the surfaces were acoustically hard; and for the lower curve, in a part of the room where the absorption was relatively high. Measurements of the reverberation times at the two points would yield the same value although the forms of the decay curves, which could be determined with a high-speed level recorder, might show characteristic differences.

The reverberation time of a room is independent of the directional characteristic of the loud speaker used as the source of sound. The character of the sound received from a loud speaker in an auditorium does, however, depend upon its directivity. By measurement of

the degree of irregularity in the transmission curve, with a particular loud speaker set in a particular way, a much better idea is obtained of the reverberant quality of the sound that is received when the loud speaker is used under the same conditions for the reproduction of speech or music. Similarly, with the recorder set to operate at slow speed, curves taken at various parts of the room with a particular speaker set-up will show the levels and the degree of balance between the high and the low frequencies of speech or music received at various parts in an auditorium.

Various other acoustic effects may be determined from transmission

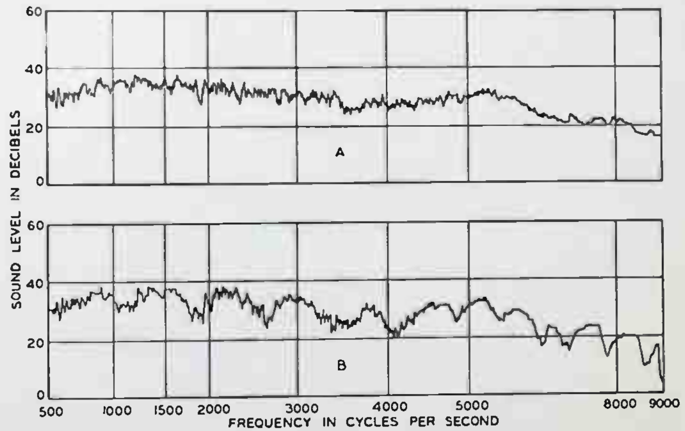

Fig. 5. Transmission *vs.* frequency characteristic between points in a room when a reflecting surface is placed near the microphone.

curves. When a microphone is placed near a reflecting surface there will be noticeable interference between the direct and the reflected sound, which may give to the reproduced sound a muffled quality. The lower curve of Fig. 5 is a transmission curve taken at a slow recorder speed when a reflecting surface was placed near the microphone. The difference in path between the direct and the reflected sound was about 15 inches. The upper curve was obtained under similar conditions, but without the reflector. A comparison of the two curves shows that the reflector produces periodic variations in the transmission, which must have a noticeable effect upon sound quality.

Although the curve of Fig. 2(B), which is representative of a room having good acoustic characteristics, is relatively smooth when compared with that of Fig. 2(*A*), the transmission characteristic is much more irregular than that of most electrical communication channels. The fact that speaking conditions are good in spite of these irregularities is to be explained partly by the nature of speech, which is never

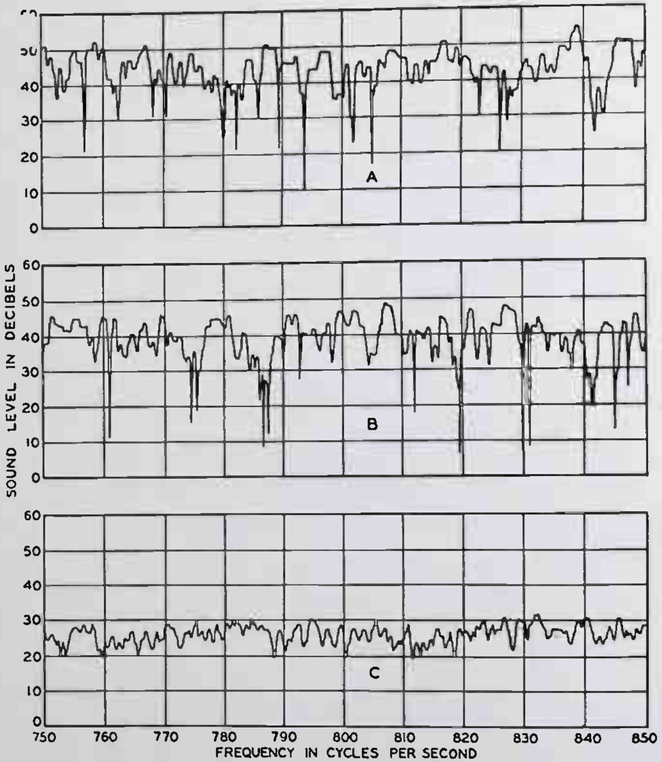

FIG. 6. Transmission *vs.* frequency characteristics under various receiving conditions: (*A*) single microphone; (*B*) six microphones connected in parallel; (*C*) six microphones, each provided with rectifier; output circuits of rectifiers connected in series.

sustained sufficiently long to set up a steady-state sound field, and partly by a phenomenon in binaural hearing. Because of the spatial separation of our ears, the sound pressures at the two ears will, in general, be different both in magnitude and in phase. If transmission measurements are made with two microphones at the receiving end connected in series and separated by the same distance as

our ears, we shall find that the transmission *vs.* frequency relation is as irregular in character as it is when the measurements are made with a single microphone, because the resultant voltage is proportional to the vector sum of the pressures at the two microphones. It is an experimental fact, however, that the loudness sensation is independent of the phase relation between the pressures at the two ears. If, for instance, the two ears are stimulated by sound pressures of the same frequency, the loudness is the same when they are in opposite phase as when they are in phase. We can, therefore, more nearly simulate binaural reception in the transmission measurements by using two microphones, each provided with a rectifier, the output circuits of which are connected in series to the level recorder; for in this case the resultant voltage will be practically independent of the phase relation of the pressures at the two microphones. The contrast between binaural and monaural reception is accentuated if the measurements are made with a greater number of microphones. The upper curve of Fig. 6 shows the transmission characteristic obtained when a single microphone is used at the receiving end. The middle curve was obtained when the single microphone was replaced by six microphones connected in parallel. The irregularities in the transmission characteristics are seen to be of the same order of magnitude in the two cases. The lower curve was obtained when each of the six microphones was provided with a rectifier and the output circuits of the rectifiers connected in series to the level recorder. This curve is obviously smoother than either of the other two, from which we conclude that in going from monaural to binaural listening in a room, the effect is similar to that produced by a reduction in the reverberation time of the room. However, comparison between Figs. 2(*B*) and 6(*C*) shows that the reduction in the irregularities obtained by increased absorption and by multiple microphones with rectifiers is not exactly of the same character: in one case the rate of decay of sound in the room is altered and in the other it is not. It is well known that when sound is reproduced by microphone and loud speaker, the reproduced sound has a more reverberant quality than when perceived directly by binaural listening in the source room. To reduce this quality it has been common and good practice to increase the damping of the source room beyond the optimal for direct listening. It is, however, not possible by this expedient to get objectively quite the same change in the reverberant quality as is achieved subjectively by binaural hearing.

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MOVING-COIL TELEPHONE RECEIVERS AND MICROPHONES

BY

E. C. WENTE and A. L. THURAS
Bell Telephone Laboratories

A DISCUSSION OF THE PRINCIPLES OF DESIGN OF A HIGH-QUALITY HEAD RECEIVER AND MICROPHONE OF THE MOVING-COIL TYPE

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MOVING-COIL TELEPHONE RECEIVERS AND MICROPHONES

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Moving-coil loud speakers are now extensively used in high quality radio-receiving sets and in talking motion picture equipment. The chief advantages of the moving coil over the moving armature driving mechanism are the absence of a static force, constancy of force-factor and electrical impedance throughout a wide frequency range, and freedom from non-linear distortion over a wide amplitude range. Because of these advantages it seems obvious that the moving coil structure can also be used profitably in head receivers and microphones where high quality is of prime importance. It has therefore been adopted in the instruments to be described, although some of the principles here formulated can conceivably be applied also to instruments with moving armatures. This paper is concerned primarily with the general principles of design. The more practical phases of the commercial design and construction of these instruments are discussed in a paper by W. C. Jones and L. W. Giles.

The moving system of a head receiver must, in general, satisfy distinctly different requirements from that of a microphone. In the actual use of the receiver a small enclosed cavity is formed between the ear and the diaphragm. If there is to be no distortion the pressure developed within this enclosure per unit of current in the receiving coil should be independent of frequency, constancy of impedance of the coil being assumed. The pressure depends not only upon the amplitude of vibration of the diaphragm, but also upon the acoustic impedance of the cavity formed by the ear and the receiver. This impedance is such, if the cavity is entirely enclosed, that at low frequencies the pressures will be very nearly proportional to the displacement of the diaphragm. At higher frequencies it is of uncertain value and varies from ear to ear, but it appears, from unpublished data obtained by L. J. Sivian on a large number of ears, that constant amplitude of motion of the diaphragm per unit current throughout the frequency range is on the average the best condition to strive for in the design of a high quality receiver. We shall therefore assume that at any frequency the amplitude of motion of the diaphragm per unit current is a correct measure of the response of the receiver. It will be assumed also that the impedance of the cavity is without effect on the displacement of the diaphragm. For the receivers to be considered this assumption introduces but little error, although the effect is not negligible in general.

The voltage generated by a moving coil in a magnetic field is proportional to the velocity; therefore the diaphragm of a uniformly sensitive microphone with a rigidly attached coil should have, at all frequencies, the same velocity per unit of pressure in the actuating sound wave. Expressed in another way, if the diaphragm has a constant effective area, the mechanical impedance (force per unit velocity) of a transmitter diaphragm should be the same at all frequencies, whereas that of the receiver should be inversely proportional to the frequency. The receiver and the microphone to be described are quite similar in design and construction, but their dynamical constants differ so as to approach these conditions of impedance.

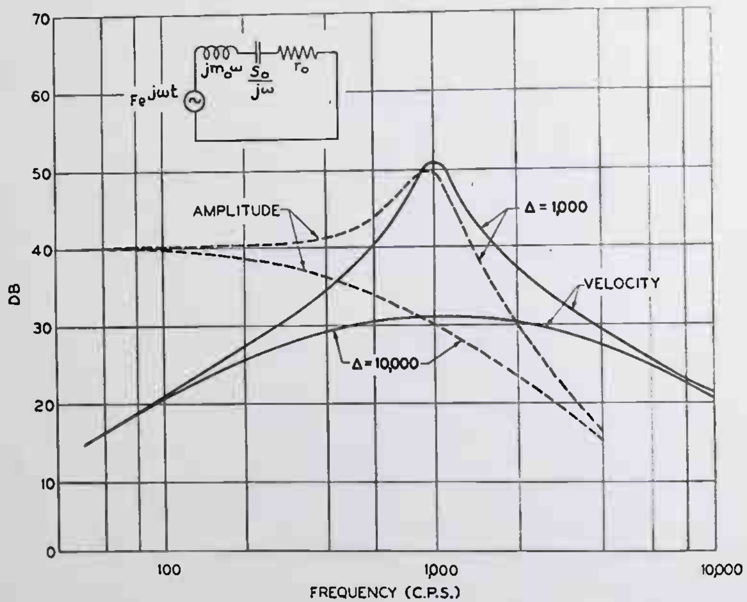


FIG. 1. *Response of a simple resonant system.*

If a receiver or microphone is constructed with a diaphragm having a single degree of freedom, the operating conditions of the diaphragm can be represented by the circuit diagram shown in Fig. 1, where m_0 is the effective mass, s_0 the stiffness, r_0 the mechanical resistance of the diaphragm, and $Fe^{j\omega t}$ the alternating force acting upon the diaphragm. The absolute value of the velocity of the diaphragm is given by

$$v = \frac{F}{m_0\left[4\Delta^2 + \left(\frac{\omega^2 - \omega_0^2}{\omega}\right)^2\right]^{1/2}}$$

and the amplitude by v/ω where $\Delta = r_0/2m_0$, the damping constant, and $\omega_0 = \sqrt{s_0/m_0} = 2\pi \times$ resonant frequency. The velocities and amplitudes for a constant force and for two different values of Δ, calculated from these expressions, are graphically represented in Fig. 1. Both the amplitude and velocity curves show wide variations in response with frequency. They indicate that for small variations in amplitude the resonant frequency must be near the upper limit of the frequencies to be transmitted and for small variations in velocity the damping constant must be high. But instruments designed on this basis would be relatively insensitive even if such conditions could be met readily in their construction.

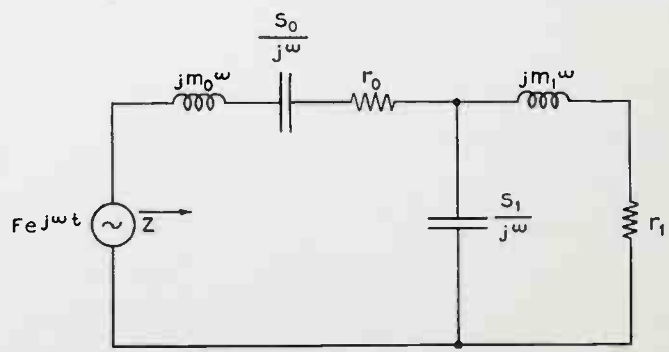
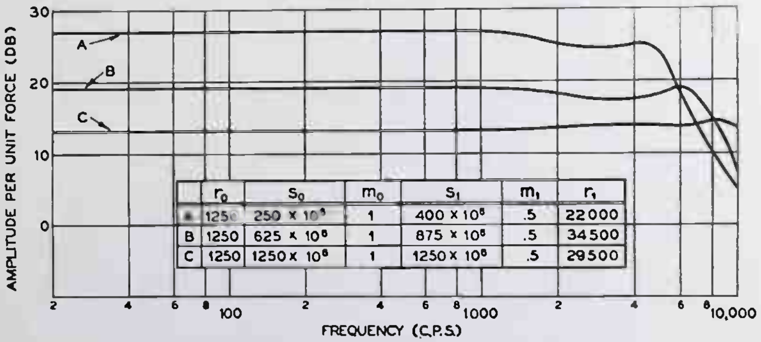


FIG. 2. *Circuit diagram for receiver or transmitter.*

In the design of electrical networks for the transmission of wide frequency bands the end is attained by the combination of more than one resonant circuit. We can advantageously resort to a similar expedient in a mechanical system by the use of a structure more complicated than one having a single degree of freedom. The diaphragm may be coupled to another mechanical or acoustical network of the proper type so as to give us the desired uniformity of response. The circuit diagram of one such mechanical network is shown in Fig. 2, where s_1 is the stiffness, m_1 the mass, and r_1 the resistance of the elements of the coupled network. The construction of a mechanical system represented by this diagram is brought out in detail in the discussion of the mechanical design of the instruments which is to follow. The actual values of the constants are to be so chosen, if possible, that the mechanical impedance, z, of the whole network is constant with frequency in the case of the transmitter and inversely proportional to the frequency for the receiver. The absolute value of this impedance, z, is $\sqrt{r^2+x^2}$ where

$$\left.\begin{aligned} r &= \frac{s_1^2 r_1}{r_1^2\omega^2 + m_1^2(\omega_1^2-\omega^2)^2} + r_0 \\ x &= \frac{s_1\omega\left[m_1^2(\omega_1^2-\omega^2) - r_1^2\right]}{r_1^2\omega^2 + m_1^2(\omega_1^2-\omega^2)^2} + m_0\omega - \frac{s_0}{\omega} \\ \omega_1^2 &= \frac{s_1}{m_1} \end{aligned}\right\} \qquad (1)$$



	r_0	s_0	m_0	s_1	m_1	r_1
A	1250	250×10^6	1	400×10^6	.5	22000
B	1250	625×10^6	1	875×10^6	.5	34500
C	1250	1250×10^6	1	1250×10^6	.5	29500

FIG. 3. *Theoretical response curves of moving coil receiver.*

THE ELECTRODYNAMIC RECEIVER

If the mechanical system of the receiver can be represented by the circuit diagram shown in Fig. 2, then, as the amplitude per unit force is a measure of the receiver response, we may calculate the product of frequency and impedance and so get a response-frequency characteristic for any specified set of values of the constants. Such characteristics are graphically shown in Fig. 3 for several sets of values. Curves of identical character but of different level would, of course, be obtained if the magnitude of each of the corresponding impedance elements were changed in the same proportion. It is seen from these curves that, theoretically at least, it is possible to obtain a uniform response over a wide frequency range. Curve c, for example, shows a variation of less than 1.5 db for frequencies up to 10,000 c.p.s. As might be expected, the wider the frequency range of uniform response the lower the sensitivity. In fact, it can be shown from equations (1) that, if the

scale of frequencies is changed by a factor, k, the relative values of the ordinates will be unchanged provided r_1 and r_0 are multipled by k, and s_1 and s_0, by k^2, but that the amplitude per unit of force at corresponding points on the curve will be changed by a factor equal to $1/k^2$. A receiver transmitting up to 10,000 c.p.s. will thus be 12 db less efficient than one transmitting equally well up to only 5000 c.p.s., the same mass and size of diaphragm being assumed.

Construction of the Receiver

The general construction of a receiver embodying the above principles is shown in Fig. 4. The central portion of the diaphragm is drawn into the form of a spherical dome to increase its rigidity. The receiving coil

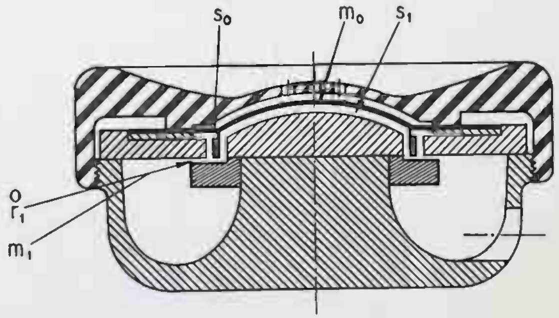


FIG. 4. *Moving coil head receiver.*

is of the self-supporting ribbon type, the construction of which has been described previously.[1] It is rigidly attached to the base of the domed portion of the diaphragm. The radial magnetic field is derived from a permanent magnet. The mass of the diaphragm plus that of the coil corresponds to m_0 in Fig. 2, the stiffness of the diaphragm to s_0 and the mechanical resistance to r_0.

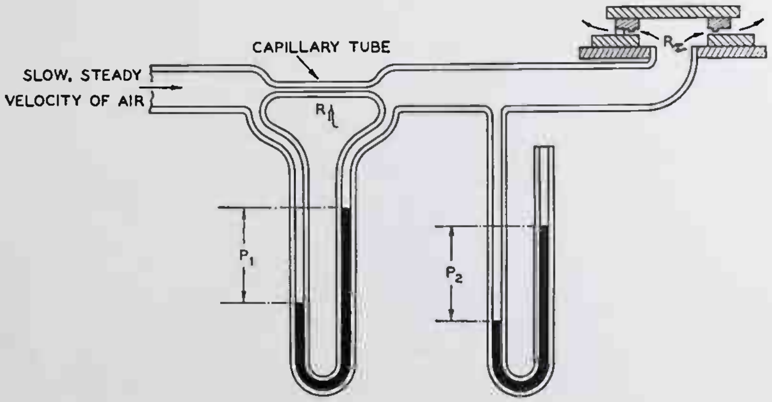
A small volume of air is completely enclosed between the diaphragm and pole-pieces save for a narrow slit at 0. The acoustic resistance[2] of a slit of this character is equal to $12\mu l/d^3w$ and the reactance, $j\,6\rho l\omega/5wd$, where μ is the viscosity of air, l the radial length, d the width, w the annular length of the slit and ρ the density of air. If the air in the chamber were incompressible a mechanical resistance and reactance would be imposed on the diaphragm by virtue of the air flow through the slit, their respective values would be equal to the acoustic resistance

[1] Bell System Technical Journal, Vol. VII, p. 144, 1928.

[2] Lamb "Hydrodynamics" 4th ed. p. 577.

and the acoustic reactance of the slit multiplied by the square of the effective area of the diaphragm. These quantities are represented by r_1 and $j\,m_1\omega$ in Fig. 2. If the slit, 0, were closed, the stiffness imposed by the air chamber on the diaphragm would be equal to $\gamma A^2 10^6/V$, where A is the effective area of the diaphragm, V the volume of air in the enclosure and γ the ratio of specific heats of air. This is the stiffness represented by s_1 in Fig. 2.

In adjusting the width of the slit to the desired value its resistance was measured experimentally. For this purpose a steady stream of air at low velocity was passed in series through the slit and a capillary tube. The pressure drop through the tube and that through the resistance was then measured with a manometer. The ratio of these values is under this condition equal to the ratio of the resistance of the tube to that of the slit. The resistance of the tube had previously been determined as a function of the pressure difference between its two ends when air was passed through it at a known steady rate. The apparatus is diagrammatically shown in Fig. 5.



R_1 = Acoustic resistance of cappillary tube
R_2 = Acoustic resistance of the annular slot
$R_2 = R_1 \times P_2/P_1$
P_1 = Pressure difference at ends of capillary tube
P_2 = Pressure difference on the two sides of the annular resistance slot

FIG. 5. *Method used to measure acoustic resistance.*

The response-frequency characteristic of the receiver was determined experimentally. For these measurements it was placed over a calibrated condenser transmitter so as to form a 15 c.c. enclosure between the receiver and the transmitter diaphragms. This space was filled with hydrogen to avoid acoustic resonance at the higher frequencies. While current from a vacuum tube oscillator was passed through the receiver coil, the voltage generated by the transmitter as well as the receiver current were measured. From these values, the calibration curve of the transmitter, and the volume of the enclosure, the amplitude of the receiver diaphragm per unit current is readily determined. Values so obtained expressed in db are plotted in Fig. 6. In the same figure are given values of the response as determined by computation of the mechanical impedance from the constants of the receiver. The ordinates were so adjusted arbitrarily as to bring the computed and observed values into coincidence at the lower frequencies. There is a general agreement between the computed and observed curves, yet the variations are larger than can be accounted for on the basis of experimental errors. It is

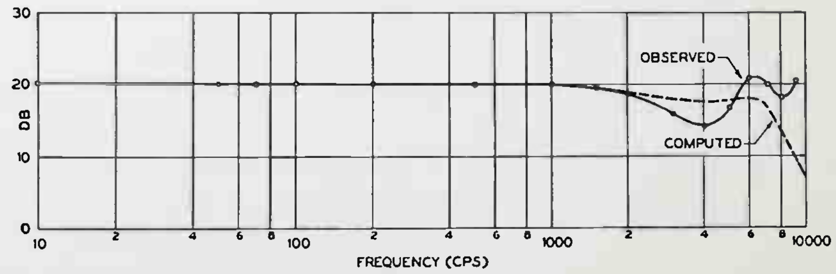


FIG. 6. *Response of moving coil receiver.*

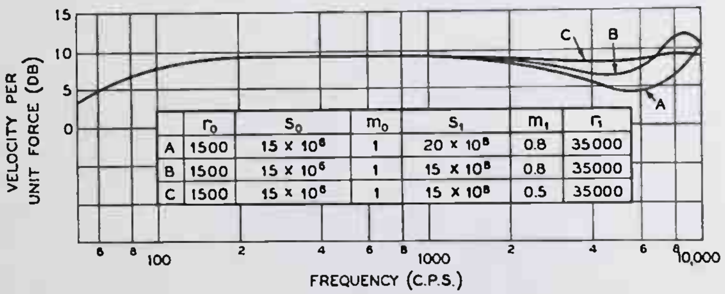
probable that the quantities used in the calculations are not strictly constant up to the higher frequencies, where the diameter of the diaphragm becomes comparable with the wave length of sound. However, except for a depression in the neighborhood of 4000 c.p.s., the measured is better than the computed characteristic.

A receiver of this general character was supplied for the Master Reference Systems for Telephone Transmission in Europe and in America where it has been in service since 1928.

The Moving Coil Microphone

It has been pointed out that in an electrodynamic microphone of high quality the diaphragm with a rigidly attached coil should have the same velocity per unit of force throughout the frequency range. If the dynamical system of the microphone is represented by the mechanical circuit of Fig. 2, this condition requires that the constants of the various elements of this circuit be so chosen that the magnitude of the impedance, z, is the same at all frequencies. It is evident that these values will differ materially from those of the high quality receiver.

In Fig. 7 the impedance expressed in db as determined by equations (1) is shown as a function of frequency for several sets of values of the constants of the impedance elements. They show how, by the proper choice of these values, a uniform response may be obtained over a wide frequency range. Curve C, for instance, shows a variation of less than 1.5 db from 200 to 10,000 c.p.s. It may be shown from equations (1) that if the scale of frequencies is changed by a factor k the form of the response curve will remain unchanged, provided r_1 and r_0 are multiplied by k, and s_1 and s_0, by k^2; but the absolute value of the velocity per unit force will be changed by the factor $1/k$ at all points on the curve. Thus, under these conditions, if the last value of the abscissae in Fig. 7 is



	r_0	S_0	m_0	S_1	m_1	r_1
A	1500	15×10^6	1	20×10^8	0.8	35000
B	1500	15×10^6	1	15×10^8	0.8	35000
C	1500	15×10^6	1	15×10^8	0.5	35000

Fig. 7.

designated as 5000 instead of 10,000 c.p.s., $k = 0.5$, the curves will remain unchanged in form but the ordinates will be raised 6 db. The form of any of the curves of Fig. 7 will, of course, not be changed if all the corresponding constants are changed proportionally, although the absolute value of the velocity per unit of force will vary inversely with the magnitude of these constants.

At zero frequency the velocity of the diaphragm per unit of force is necessarily zero. In passing to the lower frequencies a point is therefore finally reached where the response decreases appreciably. This point depends primarily upon the stiffness, s_0, of the diaphragm. A method for overcoming this loss in sensitivity at low frequencies will be discussed later.

Construction of the Microphone

A transmitter was constructed very similar in design to that of the receiver just described, but with the cap omitted in order to expose the

diaphragm to the action of sound waves. The dimensions of the various elements were changed so that the impedance of the diaphragm with its associated network should have a substantially constant value throughout a wide frequency range. The response as computed is shown in Fig. 8.

The microphone was calibrated experimentally by comparison with a calibrated condenser transmitter. For this comparison each transmitter was mounted with its face outward in an opening in the end wall of a cylindrical drum 30 cm. in diameter and 7 cm. deep. The two openings were spaced 180° with respect to the axis of the drum and on radii of 7.5 cm. Cracks between the transmitters and the wall were carefully sealed. The wall thus formed a baffle of the same general character for each transmitter. The drum was mounted on a shaft passing through its

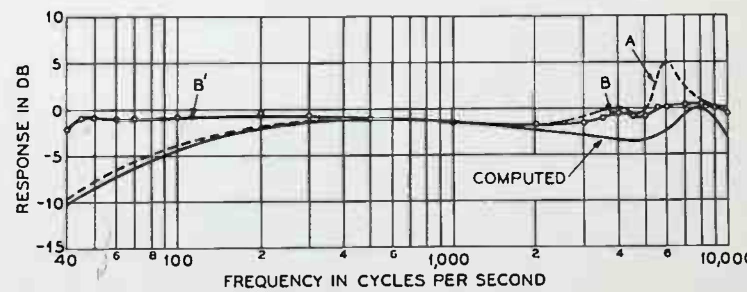


Fig. 8.

axis, about which it was rotated at a speed of 100 r.p.m. Slip rings were provided for making electrical connections to the microphones. The drum was placed in a sound field set up by a moving coil loud speaker supplied with current from a vacuum tube oscillator. The voltage generated by each transmitter was then measured with an amplifier and thermo-galvanometer. With this arrangement each transmitter passed through practically the same sound field. By virtue of the symmetrical character of the drum its rotation has very little influence on any standing wave patterns in the room. A check on the reliability of the measurements was the fact that, if the position of the loud speaker was changed, very little difference was observed in the ratio of the voltages generated even at the higher frequencies. Likewise no change was observed when the electrodynamic microphone was moved a small distance axially in its mounting. The condenser transmitter used in these tests had been calibrated by means of a thermophone, but a correction was made for the resonance due to the cavity over the face of the diaphragm, which is not measured in the thermophone calibration. The response of the transmitter as determined in this way is shown by the curve A in Fig. 8.

The disagreement between the observed and computed values at the higher frequencies is believed to be due to resonance oscillations within the air-chamber beneath the diaphragm. In order to reduce the magnitude of these oscillations the chamber was connected through a narrow slit r_3 (Fig. 9) to a small cavity formed within the central pole-piece. With this change in construction, the transmitter was again calibrated. The results obtained in this case are given in curve B of Fig. 8.

It is seen that the response of the transmitter is quite uniform over a wide frequency range, but that it decreases at the lower frequencies. This decrease can be avoided by a reduction in the stiffness, s_0, but this

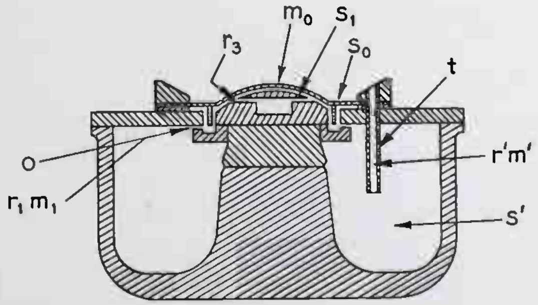


FIG. 9. *Moving coil transmitter.*

expedient has the practical disadvantage that it makes the transmitter more delicate and increases its susceptibility to mechanical vibrations. The response at these frequencies can be increased more profitably by a simple modification which increases the force on the diaphragm under the action of sound waves. If the air-space enclosed by the magnet on the rear of the diaphragm is connected with the outside air through a tube, then, under the action of sound, a pressure will be developed within this space through the tube, differing in magnitude and phase from that of the sound outside. This pressure acts on the rear of the diaphragm. Under certain circumstances the total force on the diaphragm will be increased by virtue of this pressure.

The transmitter shown in Fig. 9 is provided with a tube for performing this function. The acoustic impedance of a tube may be calculated from the formula[3]

[3] I. B. Crandall "Theory of Vibrating Systems and Sound" p. 237.

$$z = \frac{-\mu k^2 l}{\pi r^2}\left[\frac{1}{1+\dfrac{2J_0(kr)}{kJ_0(kr)}}\right] \qquad (2)$$

in which $k = \sqrt{j\mu/\rho\omega}$, l is the length and r the radius of the tube, μ the viscosity and ρ the density of air. At low frequencies, z may be represented by a resistance in series with a mass reactance, and the whole dynamical system of the transmitter by the circuit diagram of Fig. 10, in which $Fe^{j\omega t}$ is the pressure in the sound wave multiplied by the area of the diaphragm; s', the stiffness imposed upon the diaphragm by the air enclosed within the magnet, if the tube were closed; r' and m' are the acoustic resistance and mass respectively of the tube multiplied by the

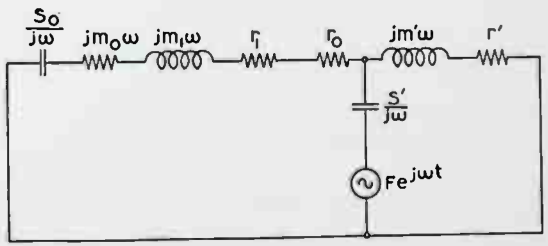


FIG. 10.

square of the area of the diaphragm. The other symbols of Fig. 10 have the same significations as before.

Substituting numerical values for the various impedances of the circuit shown in Fig. 9, and solving this circuit for the velocity of the diaphragm per unit for force, we obtained the values given by the curve B' of Fig. 8. The circles give the corresponding values obtained experimentally. The agreement between these values and those computed is within the experimental errors with which the constant of the transmitter were determined. The addition of this acoustic network has increased the response at the low frequencies so that there is no loss in sensitivity down to a frequency of 45 c.p.s., even with a diaphragm of comparatively high stiffness.

The absolute sensitivity of this transmitter is approximately 9.5×10^{-5} volts per bar. However, in practical operation a transformer is used between the transmitter and the vacuum tube of the initial stage of the amplifier. The transformer that has been used for this purpose has a voltage ratio of 100 with a variation of less than 2 db between 45 and 10,000 c.p.s. Under this condition the voltage delivered to the vacuum tube is 9.5 millivolts per bar. This value compares with approximately 3 millivolts per bar for the W. E. Co. 394 Condenser Transmitter, which was designed for maximum efficiency for frequencies up to 7,000 c.p.s. The electro-dynamic transmitter thus has a sensitivity about 10 db higher, and covers a wider frequency range.

The condenser transmitter commonly used has a cavity in front of the diaphragm. Acoustical resonance in this cavity increases the pressure on the diaphragm, which in the case of the W. E. Co. 394 Transmitter may, under certain circumstances, amount to 5 db at a frequency of 3,500 c.p.s. The transmitter here described is believed to be relatively free from this effect, as the cavity in front of the transmitter is conical and quite shallow. The diaphragm is also smaller, so that the response is uniform over a wider angle of sound incidence.

This microphone has important practical advantages over the condenser transmitter in that the amplifier may be at some distance from the transmitter without loss in efficiency and in that no polarizing voltage is required. The sensitivity of this transmitter is about 10 db higher. It is therefore better adapted for use in cases where the source of sound is at some distance from the transmitter, since with the smaller amplification required, mechanical and electrical disturbances, and amplifier noises in general, may be kept at a relatively lower level.

We are greatly indebted to Mr. T. F. Osmer, both for suggestions and for skilfully carrying out a large part of the experimental work in the development of these instruments.

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MEASUREMENT OF ACOUSTIC IMPEDANCE AND ABSORPTION COEFFICIENT

BY

E. C. WENTE and E. H. BEDELL

A DISCUSSION OF WAYS OF DETERMINING THE ACOUSTIC IMPEDANCE AND ABSORPTION COEFFICIENT OF POROUS MATERIALS FROM MEASUREMENTS ON THE STANDING WAVES IN TUBES

REPRINT OF PAPER

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The Measurement of Acoustic Impedance and the Absorption Coefficient of Porous Materials

By E. C. WENTE and E. H. BEDELL

Bell Telephone Laboratories, Incorporated

SYNOPSIS: Various ways of determining the acoustic impedance and the absorption coefficient of porous materials from measurements on the standing waves in tubes are discussed. In all cases the material under investigation is placed at one end of the tube and the sound is introduced at the other end. Values of the coefficient of absorption of a number of commonly used damping materials as obtained by one of the methods are given. Several types of built-up structures are shown to have a greater absorption coefficient for low frequency sound waves than is conveniently obtainable by a single layer of material.

THE most commonly used method of determining the sound absorption coefficient of a material is that devised by the late Professor W. C. Sabine. In this method the reverberation time of a room is measured before and after the introduction of a definite amount of the material. This method has the great merit that the values so determined usually apply to the materials precisely as they are ordinarily used in rooms for damping purposes. However, it is tedious and requires a very quiet room and large samples of the materials. A simpler scheme has been devised by H. O. Taylor,[1] in which the absorbing material is placed at one end of a tube. The coefficient of absorption is determined from a measurement of the ratio of maximum to minimum pressures of the standing waves within the tube when sound is introduced at the open end. Thus only a small sample of the material is required and with suitable apparatus the measurements can be made with great facility. In this paper several modifications of Taylor's tube method are discussed; in addition, it is shown that by a similar method it is possible to determine not only the absorption coefficient but also the acoustic impedance, a quantity which is playing an important part in present day applied acoustics.

General Theory

Consider a tube of length l, which is filled with a medium having a propagation constant $P = \alpha + i\beta$ and a characteristic acoustic im-

[1] *Phys. Rev.*, II, 1913, p. 270.

pedance[2] equal to Z_0 per unit area. At one end, O, let the velocity be uniform over the whole cross-section and equal to $\xi_1 e^{i\omega t}$. At a distance l from O let the tube be terminated by the material which is to be investigated, and the acoustic impedance of which may be rep-

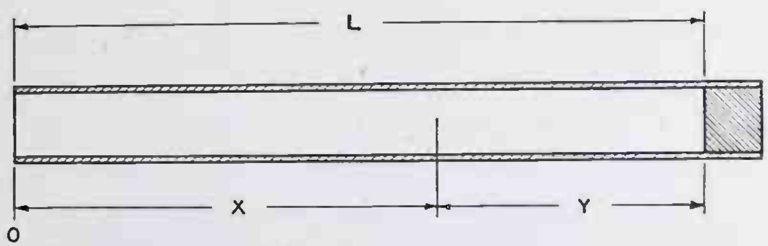

Fig. 1

resented by $Z_2 = R_2 + iX_2$ per unit area. Under these conditions, the pressure, p, at any point in the tube at a distance x from O, is by analogy with the electrical transmission line

$$p = \xi_1 e^{i\omega t}\left[\frac{Z_2 \cosh Pl + Z_0 \sinh Pl}{Z_0 \cosh Pl + Z_2 \sinh Pl} \cosh Px - \sinh Px\right] Z_0.^{3} \quad (1)$$

If there is no attenuation along the tube, we get, on dropping the time factor,

$$p = R\xi_1\left[\frac{Z_2 \cos \beta l + iR \sin \beta l}{R \cos \beta l + iZ_2 \sin \beta l} \cos \beta x - i \sin \beta x\right], \quad (2)$$

where $R = c\rho$, the product of the velocity of propagation along the tube and the density of the medium, and

$$\beta = \frac{2\pi f}{c}.$$

Equation (2) indicates numerous possible ways of determining Z_2, e.g., from the values of ξ_1 and of p at any point in the tube; from the pressures for two values of either x or l, if ξ_1 is constant; from the pressures at any point in the tube for the unknown and for a known value of Z_2; from the magnitude of p as a function of either x or l. However, we shall confine our discussion to three methods, which appear to be most practicable.

[2] The term acoustic impedance as here used may be defined as the ratio of pressure to volume velocity; the characteristic impedance is this impedance if the tube were of infinite length.

[3] J. A. Fleming, "Propagation of Electric Currents in Telephone and Telegraph Conductors," page 98; 3d Ed.

(a) Pressure Measured at Two Points in the Tube

It has already been pointed out that the impedance Z_2 can be determined if the relative phase and magnitude of the pressures at any two points in the tube are known. However, from the standpoint of convenience and precision it appears best to measure the pressures at the reflecting surface and at a point a quarter of a wave-length away. We then have at the reflecting surface $x = l$ and

$$p_2 = R\xi_1\left[\frac{R_2 + iX_2}{R \cos \beta l + iZ_2 \sin \beta l}\right],$$

and for the point $x = l - \frac{\lambda}{2} = l - \frac{\pi}{2\beta}$,

$$p_1 = R\xi_1\left[\frac{iR}{R \cos \beta l + iZ_2 \sin \beta l}\right],$$

so that

$$\frac{p_2}{p_1} = \frac{X_2 - iR_2}{R} \equiv Ae^{i\varphi}.$$

Hence

$$\left.\begin{aligned} R_2 &= -AR \sin \varphi, \\ X_2 &= AR \cos \varphi, \\ |Z_2| &= AR. \end{aligned}\right\} \quad (3)$$

If the coefficient of reflection is expressed as[4]

$$Ce^{i\psi} = \frac{Z_2 - R}{Z_2 + R}, \quad (4)$$

we get

$$C = \left[\frac{1 + 2A \sin \varphi + A^2}{1 - 2A \sin \varphi + A^2}\right]^{1/2},$$

where

$$\varphi = \tan^{-1}\frac{2A \cos \varphi}{A^2 + 1}. \quad (5)$$

The absorption coefficient, which is generally defined as the ratio of absorbed to incident power, is equal to $1 - |C|^2$.

(b) Tube of Constant Length; the Absolute Value of the Pressure Measured at Points along the Tube

The method discussed under this section is that adopted by H. O. Taylor for measuring the absorption coefficient of porous materials.

[4] I. B. Crandall, "Theory of Vibrating Systems and Sound," page 168.

For the absolute value of the pressure at any point in the tube we get from equation (2)

$$|p| = \left[\frac{R_2^2 + X_2^2 + R^2 + (R_2^2 + X_2^2 - R^2)\cos 2\beta y + 2X_2R\sin 2\beta y}{R_2^2 + X_2^2 + R^2 - (R_2^2 + X_2^2 - R^2)\cos 2\beta l - 2X_2R\sin 2\beta l}\right]^{1/2} R\xi_1, \quad (6)$$

where $y = l - x$.

$|p|$ has maximum or minimum values when

$$\tan 2\beta y = \frac{2X_2R}{X_2^2 + R_2^2 - R^2}; \quad (7)$$

for the maximum value $2\beta y$ lies in the first and for the minimum, in the third quadrant. We therefore get

$$\frac{|p|_{\max}}{|p|_{\min}} = \left[\frac{X_2^2+R_2^2+R^2+\sqrt{(X_2^2+R_2^2-R^2)^2+4X_2^2R^2}}{X_2^2+R_2^2+R^2-\sqrt{(X_2^2+R_2^2-R^2)^2+4X_2^2R^2}}\right]^{1/2} \equiv A. \quad (8)$$

Let y_1 be the value of y for which the pressure is a maximum; we then have from (7) and (8) and (4)

$$R_2 = \frac{2AR}{(A^2+1) - (A^2-1)\cos 2\beta y_1}, \quad (9)$$

$$X_2 = \frac{R(A^2-1)\sin 2\beta y_1}{(A^2+1) - (A^2-1)\cos 2\beta y}, \quad (10)$$

$$C_1 = \frac{A-1}{A+1}, \quad (11)$$

$$\Psi = 2\beta y_1.$$

The relation (11) can be derived more simply on the classical theory, as it was done by H. O. Taylor. A derivation of (11) is given by Eckhardt and Chrisler,[5] which differs from that of H. O. Taylor. From their derivation it would appear that for (11) to be valid the length of the tube should be adjusted for resonance and that the change in phase at the reflecting surface should be small. The derivation here given shows that (11) is general; it implies only that the waves be plane and that there be no dissipation of power along the tube.

[5] Scientific Paper of the Bureau of Standards, No. 526, page 56.

(*c*) *Tube of Variable Length. Pressure Measured at the Source*

The absolute value of the pressure at the driving end of the tube according to (2) is

$$|p_1| = \left[\frac{R_2^2 + X_2^2 + R^2 + (R_2^2 + X_2^2 - R^2)\cos 2\beta l + 2X_2R\sin 2\beta l}{R_2^2 + X_2^2 + R^2 + (R_2^2 + X_2^2 - R^2)\cos 2\beta l - 2X_2R\sin 2\beta l}\right]^{1/2} R\xi_1$$

and $|p_1|$ is a maximum or a minimum when

$$\tan 2\beta l = \frac{2X_2R}{R_2^2 + X_2^2 - R^2}.$$

For the maximum value $2\beta l$ lies in the first and for the minimum, in the third quadrant. We therefore have

$$\frac{|p_1|_{\max}}{|p_1|_{\min}} = \frac{X_2^2 + R_2^2 + R^2 + \sqrt{(X_2^2 + R_2^2 - R^2)^2 + 4X_2^2R^2}}{X_2^2 + R_2^2 + R^2 - \sqrt{(X_2^2 + R_2^2 - R^2)^2 + 4X_2^2R^2}} \equiv A.$$

By analogy from the equations derived in section (*b*) above, we see that

$$R_2 = \frac{2\sqrt{A}R}{(A+1) - (A-1)\cos 2\beta l_1},$$

$$X_2 = \frac{R(A-1)\sin 2\beta l_1}{(A+1) - (A-1)\cos 2\beta l_1},$$

$$C = \frac{\sqrt{A}-1}{\sqrt{A}+1},$$

$$\Psi = 2\beta l_1,$$

where l_1 is the length of the tube when p_1 has a maximum value.

Discussion of the Precision of the Methods

Of the three methods of measuring impedance discussed above, the first is undoubtedly the simplest and most convenient, if an a.c. potentiometer is available. Theoretically, in this case the impedance may be determined with a high degree of precision. However, the method presupposes that the points where the pressures are measured are exactly a quarter of a wave-length apart; a more detailed analysis shows that, if A is small, variations in this distance will have a large effect on both the ratio of the pressures and their phase difference. It therefore is necessary to keep the temperature of the tube accurately constant or else to determine the distance corresponding to a quarter of a wave-length before each measurement. A precise determination of the point a quarter of a wave-length from the reflecting surface may be made by placing a smooth metal block at the reflecting end and finding then the position in the tube at which the pressure is a minimum.

In the other two methods it is relatively less important that the temperature be maintained constant, for the ratio of pressures is affected very little by any temperature variations. In the third method, where the length of the tube is varied, the expressions for R_2 and X_2 are the same as in (*b*), except that in place of the ratio of pressures they involve the square root of this ratio. For small values of pressure ratios the precision is therefore somewhat greater. However, for high values of reflection the ratio becomes very large and great care is required in the experimental set up to prevent errors creeping into the measurements through extraneous vibrations and stray electromotive forces in the measuring circuit. The main advantage of the method in which the pressure at the source only is measured is that a short length of exploring tube is required. If measurements down to a frequency of 60 cycles are made, the tube length must be at least 8 feet. An exploring tube reaching the whole length would ordinarily introduce too much attenuation if it were of sufficiently small bore to prevent resonance effects at the lower frequencies.

Experimental Procedure

In the case of the experimental results here reported the measurements were all made by the method outlined in section (*c*), i.e., the pressures were measured at the source while the length of the tube was varied. The experimental set up is shown in Fig. 2. A piece of Shelby

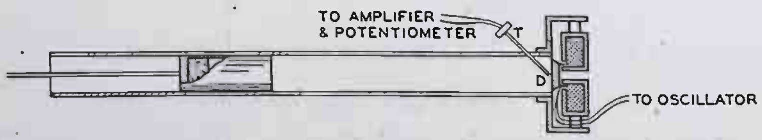

Fig. 2—Diagram of apparatus

steel tubing, 9 feet long, of 3″ internal diameter, and with 1/4″ wall, was fitted with a piston carrying the absorbing material. This piston was made up of a brass tube one foot long with a wall 1/64″ thick, the far end of which was closed with a one-inch brass block. To insure the propagation of plane waves and a constant velocity at the source, the diaphragm at D had a diameter of $2\frac{7}{8}''$, and a mass of about 100 grams. This was driven with a coil 2″ in diameter situated in a radial magnetic field. The annular gap between the edge of the diaphragm

and the interior of the tube was closed by a flexible piece of leather. To prevent vibrations of the magnet from getting to the tube, the magnet was held in position by flexible supports. The exploring tube t was about 5″ long with a 1/16″ bore which led to the transmitter, T. The voltages generated by the transmitter were measured with an amplifier and an a.c. potentiometer. The potentiometer was used because with it small voltages can be measured and errors due to harmonics are avoided. The proper functioning of the apparatus was determined by measuring the coefficient of reflection with no absorbing material in the piston. Theoretically the reflection should then be practically 100 per cent. The pressure ratios that were actually observed were of the order of 12,000 which corresponds to a reflection coefficient of 98 per cent. Evidently some extraneous pressures or voltages were still present. However, no attempt was made to reduce these further as the materials tested had a reflection coefficient considerably less than this value.

Experimental Results

A brief study was made of the absorption of hair felt, as there is an appreciable variation in the data given by various investigators on the absorption frequency characteristic of felts of presumably the same

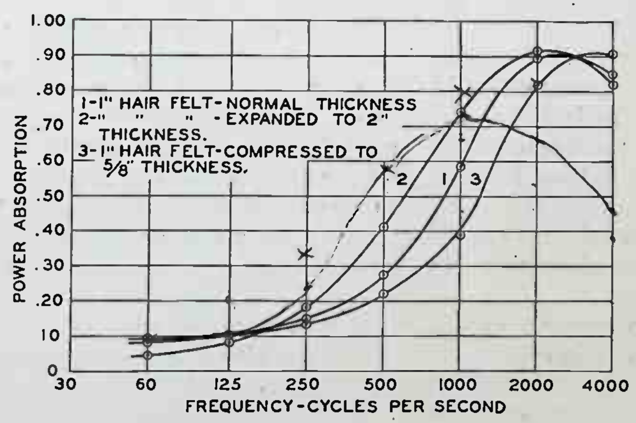

Fig. 3—Power absorbed by hair felt

type. After measurements on several samples it was evident that concordant results could not be expected as the absorption varied considerably with the packing of the felt. This point is illustrated by the curves shown in Fig. 3. These curves were all obtained on the same piece of hair felt but with different degrees of packing. It is thus evident that a felt which has become loosened by handling may have an absorption frequency characteristic quite unlike that of a new piece.

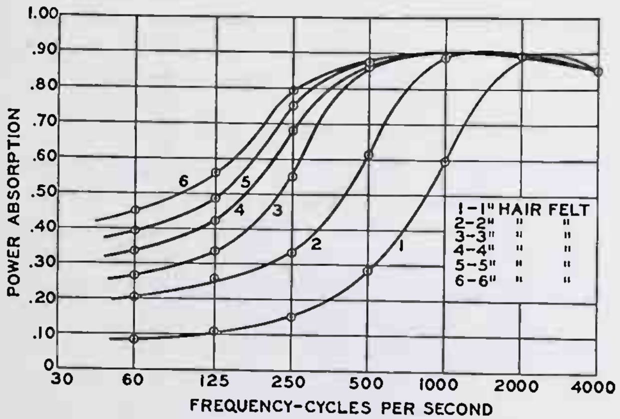

Fig. 4—Power absorbed by hair felt

In Fig. 4 are given the absorption coefficients for various thicknesses of hair felt. These values are in general agreement with those obtained by the reverberation method according to published results. Exact agreement is not to be expected, for the values here given apply only to sound waves having a perpendicular incidence on materials solidly backed by a hard surface. When the materials are applied in a room, the support is often more flexible and the absorption is partly due to inelastic bending. However, the agreement between the sets of values is sufficiently good to show that the results obtained by the simpler tube method may be used to get a good approximation to the values of absorption of the materials when applied in rooms for damping purposes.

Measurements have been made on a large number of porous materials. Although most of these materials are very good absorbers at the higher frequencies, none of them were found to be very efficient in the lower frequency region. Uniform absorption over most of the frequency range was found only in materials which are relatively inefficient absorbers. High absorption at the lower frequencies was obtained only when the thickness of the material was greatly increased. This fact is typically illustrated by the curves of absorption for hair felt given in Fig. 4.

When a sound wave of low frequency is reflected from a wall covered

TABLE I

Absorption Coefficients for Various Frequencies

Frequency c.p.s.

	60	125	250	500	1000	2000	4000
1″ Acoustic tile	.07	.08	.11	.18	.48	.76	.47
½″ Asbestos hair felt	.08	.10	.15	.24	.49	.84	.66
Axminster rug	.07	.11	.14	.20	.33	.52	.82
Felted wood fibre	.08	.12	.18	.33	.67	.92	.91
½″ Building board	.13	.14	.15	.17	.20	.26	.29
Flax wool	.07	.09	.18	.48	.73	.50	.33
Structure No. 1	.12	.18	.36	.71	.79	.82	.85
" " 2	.16	.24	.46	.77	.92	.89	.85
" " 3	.23	.37	.62	.88	.91	.78	.84
" " 4	.19	.28	.51	.81	.92	.90	.84
" " 5	.15	.25	.44	.75	.77	.71	.80
" " 6	.22	.41	.87	.74	.81	.59	.83
" " 7	.17	.39	.82	.94	.92	.91	.85
" " 8	.24	.39	.83	.82	.64	.59	.80
" " 9	.22	.37	.79	.91	.82	.89	.86
" " 10	.30	.55	.92	.69	.83	.86	.86

Structure No. 1	Fiber building board—no air space—felt
" " 2	Fiber building board—1″ air space—felt
" " 3	Fiber building board—2″ air space—felt
" " 4	Fiber building board—no air space—2″ felt
" " 5	Fiber building board—1″ air space—Fiber building board
" " 6	Fiber building board—1″ air space—Fiber building board—1″ air space—Fiber building board
" " 7	Fiber building board—1″ air space—felt—1″ air space—felt
" " 8	Fiber building board—1″ air space—felt—1″ air space—Fiber building board
" " 9	Fiber building board—1″ air space—Fiber building board—1″ air space—felt
" " 10	Fiber building board—1″ air space—Fiber building board—1″ air space—Fiber building board—felt

with a porous material, the velocity of the air particles near the reflecting surface is small and hence there can be but little absorption. We may look at the phenomenon of reflection in still another way. In order to have a small coefficient of reflection the mechanical impedance of the wall per unit area should, as nearly as possible, be equal to the acoustic impedance of the air per unit area. The reason for the high reflection at low frequencies by a rigid wall covered with a porous material lies in its high stiffness reactance. At a given frequency this reactance can be compensated by loading the air near the reflecting surface. This may be accomplished in various ways. One of these ways is to place at a short distance from the wall a second wall which is porous or perforated. This arrangement has the effect of covering the wall with a multiplicity of resonators, which may be given any desired resonance frequency by properly proportioning the size, length and number of perforations and the spacing of the walls. The surface of the walls forming the air space should be absorbing or else the space should be provided with absorbing material.

To get a wider absorption band two or more perforated walls with proper spacing may be used, as this arrangement is equivalent to an aggregate of multiple resonators. The values of absorption coefficients of a number of structures of this type are given in the accompanying table. The measurements refer to sound which is incident from right to left as the structures are given in the table. The building board referred to in the table is a commercial type of insulating-board one inch thick with 400 1/4 inch by 3/4 inch holes per square foot. The felt in all cases is one-inch hair felt. These values show that relatively high absorption may be obtained at low as well as at high frequencies without an excessive amount of absorbing material. The use of combinations of absorbing materials, such as are given in the table, offers the advantage that more uniform damping at all frequencies can be obtained, and the degree of damping can be readily controlled by covering the proper area of surface. These two factors have become increasingly important in studio and auditorium design, with improved technique in recording and reproducing speech and music.

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A SURVEY OF THE MORE RECENT ELECTRICAL DEVICES
USED IN THE STUDY OF AIR-BORNE
SOUND WAVES

Presented before

ACOUSTICAL SOCIETY OF AMERICA
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Acoustical Instruments

By E. C. WENTE
Bell Telephone Laboratories

Previous to the development of amplifiers most of the instruments used in acoustical research depended for their operation upon purely mechanical principles. This paper includes a brief survey of such of these instruments as are still of interest in connection with the investigation of technical or research problems in acoustics, but it deals primarily with the more recent electrical devices used in the study of air-borne sound waves.

The limitations and fields of application of various electrical instruments, including microphones, particularly adapted to definite types of acoustic measurements, are discussed.

MEASUREMENTS in acoustics may be said to date from the fifth century B.C., when Pythagoras observed that the lengths of strings giving the fifth, the fourth and the octave had the ratios 6 : 4 : 3, but no further really significant quantitative acoustic measurements were reported until the 17th centry when the frequencies of vibration of the notes in the musical scale were determined by Mersenne.[1] The first systematic treatise on experimental acoustics was published by Chladni [2] whose work on the vibration of plates and diaphragms is well known. With respect to the development of present day acoustical instruments the most outstanding contribution of the last century was the application of diaphragms for receiving sound waves by Scott and Koenig. Such diaphragms not only are used in most of these instruments, but also form an important element in two notable inventions of the last century, the telephone and the phonograph.

One of the chief functions of an acoustic diaphragm is to translate the extremely small pressures of sound waves into comparatively large corresponding forces, but a diaphragm cannot deliver more power to a system than it absorbs from the sound field. Telephony over comparatively long distances was made possible by the invention of the carbon microphone, an instrument which is capable of translating the small powers of acoustic diaphragms into relatively much larger electrical powers. This microphone, while of great commercial utility, was, for a number of reasons, unsuited for most quantitative acoustic measurements. Practically all shackles were removed from

[1] *Harmonie Universelle* (1636).
[2] *Die Akustik* (1802).

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the designer of acoustical instruments some twenty years ago through the invention of the vacuum tube telephone amplifier. Previously his chief concern lay in making a device sufficiently sensitive to give a measurable response; now sensitivity became of secondary importance, and attention could be focused on the design of instruments which should be capable of performing their function without distortion.

As a result of this invention, instruments depending upon an amplifier for their utility have come so to dominate the field of acoustic measurements that we might easily be led to disregard all others. A number of such other instruments have, however, in recent years been brought to a high state of development which are peculiarly suited either for the calibration of other devices, or for the study of certain special problems. This paper attempts to give a brief critical survey of the various types of acoustical instruments which at the present time are finding applications in technical fields and in acoustical research studies.

General Principles

The Rayleigh Disc

That under certain conditions a torque is exerted on a thin disc suspended by a fine fibre in a stream of air was first observed by the late Lord Rayleigh,[3] who recognized in this phenomenon a means for measuring the intensity of sound.

The following quantitative relationship between the torque and the stream velocity was derived by W. Koenig:[4]

$$\tau = \frac{4}{3}\rho_0 r^3 u^2 \sin 2\theta,$$

where ρ_0 is the mean density of the medium, r the radius of the disc, u the stream velocity, and θ the angle between the undisturbed stream and the normal to the disc. The assumptions underlying the derivation of this formula are that the fluid is incompressible, that the disc is an infinitely thin ellipsoid and that there are no forces due to viscosity or to discontinuities of flow at the edges of the disc; i.e., the velocities are derivable from a potential. In view of these assumptions how far may we rely on the above formula in applying it to a suspended plane flat disc as commonly used for measuring the particle velocity of a sound wave, where none of these assumptions are strictly fulfilled? In most acoustical problems it is perfectly safe to assume a potential field as the forces due to viscosity and eddies are of the

[3] *Phil. Mag.* 14, 186 (1882).
[4] *Wied. Ann.* 43, 43 (1891).

second order. With respect to the torque on the Rayleigh disc it is not so obvious that such forces may be neglected as the potential torque is itself of the second order. The effects of discontinuities in the flow at the edges and of viscosity have not been determined theoretically. To what extent these are negligible can be found out only by experiment. Strictly speaking we therefore cannot regard the Rayleigh disc as an absolute means for determining sound intensities, as is often implied. A number of experiments have been carried out to determine the accuracy of the Koenig formula. Koenig himself, in a subsequent paper,[5] made an estimate of the effect of the discontinuous flow at the edges and reported measurements of the torque exerted on a flat disc when placed in a steady stream of air of known velocity. As a result of these studies he came to the conclusion that the application of his simplified formula to the Rayleigh disc did not provide a reliable and simple method for measuring the absolute value of sound intensity. He felt that the effect of viscosity would probably also have to be taken into account. However, Koenig's experiments were made under difficult conditions and it is possible that his measurements were affected by eddies in the air stream.

Greater confidence in the accuracy of Koenig's formula is derived from the experiments of Zernow.[6] Zernow experimented with both thin true ellipsoids and flat discs; these were placed in a box attached to one prong of a tuning fork, the motion of which was observed microscopically. The tuning fork was driven at a frequency of 92 c.p.s. and the relation between the amplitude of motion and the resulting deflection of the disc was determined. The values so found for the ellipsoids agreed remarkably closely with those computed by the formula. For the discs the agreement was within about 10 per cent. On the basis of the values so found Zernow proposed an empirical correction factor which reduces to unity for infinitely thin discs. Barnes and West,[7] using thinner discs, made measurements similar to those of Zernow. They found almost perfect agreement between the experimental and the theoretical values. They were able to show also, by measurements made at audio frequencies with discs of different diameters, that the torque varied as the cube of the diameter, provided first, that the diameters did not exceed 1/5 wave-length, and second, that the discs were sufficiently rigid to be free from resonant vibrations at the measuring frequency. Mallet and Dutton[8] found that the torque was proportional to the square of the velocity up to

[5] *Wied. Ann.* 50, 639 (1893).
[6] *Ann. d. Physik* 26, 79 (1908).
[7] *Jour. I.E.E.* 65, 871 (1927).
[8] *Jour. I.E.E.* 63, 502 (1925).

velocities of 5 cm. per second. We might, however, expect that a torque resulting from discontinuous stream flow at the edges would be governed by a similar law.

No direct tests have been reported on the accuracy of the coefficient in Koenig's formula above 92 c.p.s., at which Zernow's measurements were made. However, sound intensities determined by the Rayleigh disc through the application of Koenig's formula have been found to be in good agreement with those determined with a microphone calibrated by other independent means. All the tests of the formula have been made with plane or spherical sound waves of moderate intensity. This fact should be borne in mind in order to guard against the use of the device under conditions where the formula may not be applicable. Such may be the case, for instance, where the sound intensity is very high. Measurements in non-uniform sound fields recently made by Kotowski [9] showed quite anomalous effects; in some cases the deflection was even in a direction opposite to that expected.

One great disadvantage of the Rayleigh disc method of measuring sound intensity, as ordinarily applied, is the fact that the disc will deflect under the action of a steady air stream. As the stream velocities in a sound wave are in any case quite small, circulating air currents may easily produce comparable deflections unless the instrument is well shielded therefrom. Under carefully controlled conditions measurements can be accurately made at sound intensities corresponding to pressures as low as one bar.

The effect of circulating air currents is greatly reduced in the method of measurement with the Rayleigh disc adopted by Sivian.[10] In this method the intensity of the sound to be measured is modulated at the source at a frequency of about 0.4 cycle per second. The disc with its suspension is proportioned so that its natural frequency is equal to this modulating frequency. The disc will then oscillate under the action of the modulated sound wave at an amplitude proportional to the square of the velocity. As circulating air currents generally have components lying below the modulating frequency they will have but little effect on the amplitude of the oscillations of the disc.

Determination of Intensity from Static Pressure Measurements

Another purely mechanical means for measuring sound intensity in absolute terms is based upon the fact that when radiant energy falls on a reflecting surface a static pressure is exerted on this surface, which in the case of sound is equal to [11] $((\gamma + 1)/2)I/c$, where I is the

[9] *E.N.T.* 9, 404 (1932).
[10] *Phil. Mag.* 5, 615 (1928).
[11] Lord Rayleigh, *Phil. Mag.* 10, 366 (1905).

intensity and c is the velocity of sound. A disc which just clears the opening in a plane baffle wall is attached to one arm of a torsion balance. From the deflection of the balance when sound falls at perpendicular incidence on the disc the radiation pressure and hence the intensity of the sound may be determined. This method has been successfully used in experiments with supersonic waves. At these high frequencies the diameter of the disc may be made a large fraction of a wavelength and the baffle may be omitted. At audio frequencies the necessity of using a baffle is a distinct handicap to this method.

Since the relation between pressure and condensation in air is not strictly linear a sound wave will, under certain circumstances, produce a change in static pressure. For a plane wave this has been shown by Thuras, Jenkins and O'Neil [12] to be equal to $-((\gamma + 1)/4) \times I/c$, where I is the sound intensity. Eichenwald [13] has suggested that a measurement of this pressure should provide a means for determining the absolute value of the sound intensity. Such increments in static pressure can, however, exist only when equalization by air flow to regions of normal pressure is precluded, a condition not easily established in practice.

Acoustic Valve

An extremely simple device for measuring sound intensities was devised by Kundt.[14] One end of a tube, which is placed in the sound field, is terminated by a valve which is so delicate that it will close during the negative and open during the positive half of the pressure cycle of the sound wave. The other end of the tube is terminated by a manometer. With perfect operation of the valve the sound wave will force air into the tube until the pressure indicated by the manometer is approximately equal to the maximum pressure in the sound wave. Recently Eisenhour and Tyzzer [15] have developed a sound meter operating on this principle. It is provided with an ingenious type of sensitive manometer with which the pressures are indicated on a dial. It has a fairly uniform sensitivity up to 2,000 c.p.s. The construction of the valve used in this meter is not disclosed in the literature. However, Ribbentrop [16] recently has described a similar sound meter in which the valve consists of the wing of a house-fly placed over an opening. It is stated that the instrument is capable of giving reliable measurements for sound pressures above 70 bars.

[12] *Jour. Acous. Soc. Amer.*, January, 1935.
[13] *Rend. Sem. Mat. e Fisico d. Milano*, Vol. 6 (1932).
[14] *Ann. d. Physik* 134, 568 (1868).
[15] *Jour. Franklin Inst.* 208, 397 (1929).
[16] *Zs. f. Tech. Phys.* 13, 396 (1932).

Measurement of Periodic Changes in Density

As the optical index of refraction of an elastic medium depends upon the density, it is possible to measure sound by letting one of the paths of the light beams of an interferometer pass through the sound field while the other is shielded therefrom. The interference fringes of the interferometer will be displaced periodically in synchronism with the periodic variations in density of the wave. This method was first used by Boltzmann and Toepler [17] who in this manner observed the rather large variations in density within a sounding organ pipe. This method has the advantage that the measurements are independent of frequency but it is not very sensitive and at best is rather cumbersome. An interesting modification [18] of this method has recently been applied in measurements of high-frequency sound waves in liquids. At these high frequencies the wave-lengths are so small that the spatially periodic variations of the density of the medium can act as a diffraction grating for light waves. This phenomenon has provided a neat means of picturing the propagation of high-frequency sound waves in liquids.[19]

Instruments Employing Diaphragms and Optical Magnification

In the phonautograph of Scott (1857) a circular diaphragm is actuated by sound waves and the motion is recorded on a moving strip of smoked paper by a stylus attached to the center of the diaphragm. The recorded amplitudes are no greater than the actual amplitudes of motion of the diaphragm which, except at the resonance frequency or for very intense sounds, are so small that they cannot be accurately determined from the record. Small motions can be observed and recorded if the stylus is replaced by an optical lever. This arrangement in various forms has been used in the past by a number of investigators. It reached its highest state of development in the well-known phonodeik of D. C. Miller.[20] In this instrument a horn is used for increasing the sound pressure acting on the diaphragm, the motion of which is magnified in some forms of the instrument by as much as 40,000 times. By refinements in mechanical design and construction a remarkably uniform sensitivity was achieved.

Microphones

The instruments discussed so far operate without the benefit of electric-current amplifiers. The important role that these amplifiers

[17] *Pogg. Ann.* 141, 321 (1870).
[18] Debye and Sears, *Proc. Nat. Acad. Sci.* 18, 409 (1932). Lucas and Biquard, *Jour. de Physique et le Radium* 3, 464 (1932).
[19] R. Baer and E. Meyer, *Phys. Zeits.* 34, 393 (1935).
[20] *Science of Musical Sounds*, The Macmillan Company (1922).

have played in recent developments of acoustic instruments has already been indicated. To apply such amplifying means we must first of all have a device to convert sound power into electrical power. By far the most important instrument of this class is the microphone, which is a device that translates sound into corresponding electrical currents. When a medium is traversed by a sound wave it undergoes periodic variations in pressure, density, temperature and particle velocity. A device which translates any one of these variations into corresponding electrical currents may be classified as a microphone.

The great utility of the carbon microphone rests upon the fact that it in itself functions as an amplifier, i.e., the electrical power generated is greater than that absorbed from the actuating sound wave. The carbon microphone, however, has not been widely used for acoustic measurements, lacking the requisite stability and constancy. After amplifiers became available high sensitivity was no longer so important. It became possible to develop microphones in which high sensitivity was a subordinate property but which were stable and constant and relatively free from distortion.

The sensitivity of a microphone as a function of the frequency can usually not be easily determined from its physical constants. It must, therefore, be calibrated to be useful for general acoustic measurements. Such calibrations are commonly made in terms either of the voltage generated per unit of pressure acting on the instrument, or of the voltage per unit of the pressure obtaining in a plane progressive sound wave before the microphone is placed in the sound field. The former is referred to as a pressure and the latter as a free field calibration. Very complete discussions of the various methods of effecting such calibrations have been given by L. J. Sivian [21] and by S. Ballantine.[22] Unless the dimensions are small compared with the wave-length the microphone will diffract the sound waves and the pressure on the diaphragm will not be the same as that of the undisturbed sound field; for example, at normal incidence and at frequencies for which the wave-length is small compared with the diameter of the microphone the pressure will be doubled. The diffraction effect exhibits itself, particularly in a variation in the response-frequency characteristic with angle of incidence of the sound wave, generally in not an easily predetermined manner. If the form of the instrument is that of a sphere it is possible to determine this variation with angle of incidence theoretically. Ballantine [23] and also Oliver [24] have, there-

[21] *Bell Sys. Tech. Jour.* X, 96 (1931).
[22] *Jour. Acous. Soc. Amer.* 3, 329 (1932).
[23] *Phys. Rev.* 32, 988 (1928).
[24] *Jour. Sci. Inst.* 7, 113 (1930).

fore, worked with instruments of this form. In any type of microphone diffraction effects can be entirely eliminated only by making the dimensions small compared with the wave-length.

The calibration of a microphone for a particular sound field may be carried out by measuring the undisturbed field with a device which is small compared with the wave-length and then noting the response of the instrument when placed in this field. This kind of calibration, when made in a nearly plane progressive wave, is referred to as a free field calibration. For the standard measuring instrument a Rayleigh disc is commonly used. This calibration is then applicable only for cases where we have substantially this type of sound field, i.e., when the microphone is at some distance from the source and all the sound is received by direct transmission. Where this condition is not fulfilled, the free field calibration is no true indication of the performance; for instance, when an instrument is used as a close talking microphone our experience indicates that in some cases at least an instrument having a flat characteristic, as obtained by a pressure calibration, delivers a voltage having frequency components of more nearly the same relative intensity as that in the voice when no microphone is near the mouth than does a microphone having a flat characteristic as given by a free field calibration. To eliminate diffraction effects a number of investigators have constructed microphones of small size, to some of which reference will be made in subsequent sections. Where it is necessary to make measurements with an extremely small instrument, such as in the exploration of the sound field within conduits and horns, the most satisfactory method of procedure is to use a small tube leading to a chamber closed over the diaphragm of a larger microphone.[25] The disadvantage of this arrangement is the fact that the loss in pressure through such tubes increases rapidly with frequency, so that at high frequencies it is necessary to work with high sound intensities or use uncomfortably high gain amplifiers. In working with single frequencies a great advantage in ease of measurement can be gained by the use of band-pass filters.

Pressure Microphones

Although microphones may conceivably be designed to translate directly the periodic variations of pressure, temperature, density, or particle velocity of a sound wave into corresponding electrical voltages, it is convenient to divide them into two classes: pressure microphones and velocity microphones, since the first three of the above characteristics of sound waves are proportional in any type of sound field.

[25] Sell, *Wiss. Ver. d. Siemens-Konz.* 2, 353 (1922).

Condenser Microphone

One of the first so-called high-quality microphones developed for use with amplifiers was of the condenser type. This is in principle one of the simplest of all microphones. It consists essentially only of two parallel insulated plates, one of them fixed and the other movable under the action of the alternating pressure of the sound wave. When these plates are connected in series with a resistance and a battery an alternating current will flow in this circuit in accordance with the variations in capacitance between the two plates. The resulting potential variations across the resistance are impressed on the grid of a vacuum tube.

A different method of using the condenser microphone has been described by Riegger.[26] The microphone is made a part of the capacitance element of a high-frequency electric oscillator. The frequency of the oscillations is thus modulated in accordance with the sound pressure acting on the diaphragm. If the modulated current is transmitted through a circuit, the transmission of which varies linearly with the frequency, in series with a linear rectifier, the output current of the rectifier will correspond to the sound pressure.

The condenser microphone as commonly used is of a size such that at the higher acoustic frequencies it will distort the sound field. The pressure and free field calibrations begin to diverge from each other at about 1000 c.p.s. To eliminate this distorting effect a number of investigators[27] have developed miniature condenser microphones for laboratory use. Generally such instruments have been designed at a sacrifice in sensitivity and uniformity of response. The small size microphone developed by Harrison and Flanders, however, has a remarkably flat response frequency characteristic and a sensitivity comparable with that of the larger instrument. Still smaller condenser microphones have been constructed but at a sacrifice in sensitivity.

At this point it may be of interest to give an example which illustrates the great advantage that the vacuum tube amplifier has given us in the design of sound measuring instruments. With an amplifier having a uniform amplification from 50 to 10,000 cycles, it is possible to measure, under favorable conditions, voltages as low as 1 microvolt. The amplitude of motion of the diaphragm of a common form of condenser transmitter delivering this voltage is about 10^{-11} cm., or about 1/1000 of an Angstrom. This illustrates the extremely small amount of motion that has to be imparted to the moving element of the

[26] *Wiss. Ver. Siemens-Konz.* 3, 2, 67–100 (1924).
[27] K. Hall, *Jour. Acous. Soc. Amer.* 4, 83 (1932). Harrison and Flanders, *Bell. Sys. Tech. Jour.* XI, 451 (1932).

measuring instrument. Not even with an optical interferometer could we hope to evaluate displacements so small.

Moving Coil Microphone

The condenser microphone has inherently a high electrical impedance, so high in fact that any attempt to connect the microphone to an amplifier by leads of appreciable length results in a loss of voltage. To avoid this loss an amplifier of at least one stage has generally been placed in close connection with the microphone. However, since the input impedance of a vacuum tube is also high, the microphone can be connected to it without the use of an impedance transformer, a distinct advantage at the time when transformers of good frequency characteristic were not available. During the last few years, through the development of new magnetic materials and advances in design, it has been possible to build transformers having a substantially uniform response over the whole acoustic frequency range. This development has made it possible to design microphones operating on electromagnetic principles, which have a good response-frequency characteristic and a greater sensitivity than the condenser microphone. They have an important advantage over the condenser microphone in that, because of their relatively low and constant impedance, they may be connected to the amplifier by a relatively long cable without appreciable loss. One such instrument [28] is shown diagrammatically in Fig. 1.

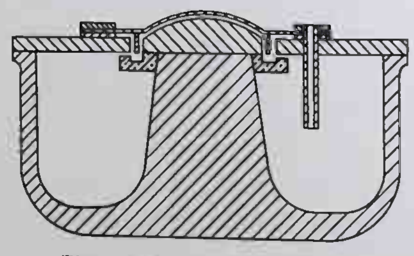

Fig. 1—Moving coil microphone.

The diaphragm has attached to it a coil which lies within a radial magnetic field. As the voltage generated by an axial motion of the coil is proportional to the velocity, if the same voltage is to be generated at all frequencies under a given sound pressure, the impedance of the moving element must be independent of frequency. This type of impedance characteristic over a wide frequency range is obtained by properly proportioned air chambers and resistances in back of the

[28] *Jour. Acous. Soc. Amer.* 3, 44 (1931).

diaphragm. This microphone, when provided with a coil having an electrical resistance of 20 ohms, will generate 10^{-4} volts per bar of sound pressure. The smallest voltage that can be measured at the terminals of a resistance is limited by the voltage due to thermal agitation of the electrons,[29] which under normal conditions and for a frequency band of 15,000 c.p.s. is equal to 7×10^{-8} volts for a resistance of 20 ohms. Hence the smallest pressure that it is possible to measure with this microphone is about 7×10^{-4} bars. However, over a narrow band of frequencies, or at a single frequency, measurements may be made down to still lower pressures if the circuit is provided with a band-pass filter. The sensitivity of this instrument is higher than that of any other microphone of comparable frequency range at present available. In evaluating some of the other microphone principles we shall, therefore, use its sensitivity as a reference, without meaning to imply that sensitivity is the sole criterion of the merit of a microphone. There is also an upper limit to the sound intensities that may be measured with this instrument. This is governed by the maximum amplitude of excursion that the diaphragm can make without the generation of appreciable harmonics. The upper and lower limits at the various frequencies are shown by the curves in Fig. 2.

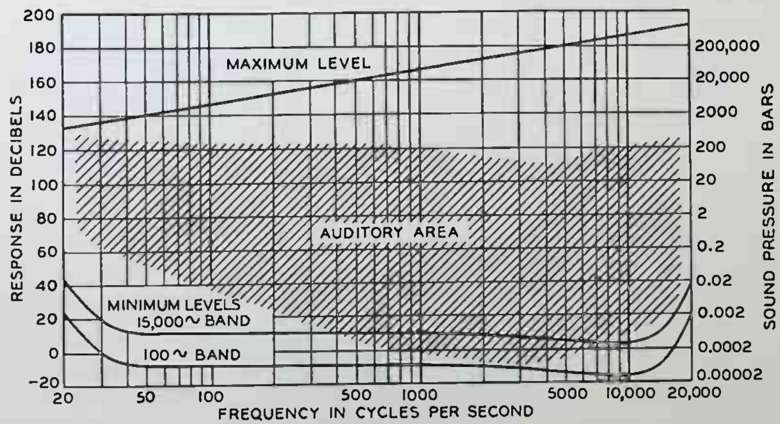

Fig. 2—Operating range of moving coil microphone.

The upper limit is taken as the pressure at which the higher harmonic components of the voltage are equal to 3 per cent of the fundamental. The lower limit represents the pressure at which the signal voltage is just equal to the voltage of thermal agitation. For comparison the

[29] J. B. Johnson, *Phys. Rev.* 32, 97 (1928).

corresponding auditory range is indicated by the cross-sectioned area. It will be noted that when operated at its full frequency range even this relatively sensitive microphone is incapable of translating practically sound of intensities as low as the ear can hear.

This instrument, which in a similar form is used as a commercial microphone, is several inches in diameter and so is not without effect on the sound field. Where a certain amount of operating range and sensitivity may be sacrificed, as in many acoustic measurements, it is possible to construct this instrument in a much smaller form.

Capillary and Magnetostriction Microphones

Besides the electrostatic and electromagnetic methods of translating the mechanical pressures of a sound wave into corresponding electrical potentials, there are other electromechanical phenomena which may be applied for the purpose. Outstanding among these are the capillary electrometer, magnetostriction, and piezoelectric action.

When a potential is applied at the interface between an electrolyte and mercury the surface tension is changed. If the mercury is in a capillary tube the change in surface tension will result in a change in position of the mercury; conversely when a force tending to move the surface is applied, there will be a resulting change in potential across the interface. This phenomenon has been applied in the design of microphones. One form of construction of such an instrument is shown in Fig. 3, taken from a paper by Latour.[30] The instrument

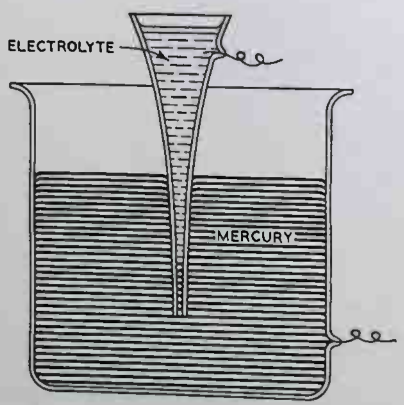

Fig. 3—Latour capillary microphone.

appears to have been used but little up to the present time and there seems to be very little in the literature regarding its performance.

[30] *Compt. Rend.* 186, 223 (1928).

Magnetostriction also has found little application in microphones for air-borne waves at audio frequencies, although this principle has been applied with notable success in both the generation and detection of ultra-audio waves by G. W. Pierce and in the generation of audio frequency waves of high intensity in liquids.[31]

Piezoelectric Microphones

The application of piezoelectric action in the construction of acoustic microphones was first made by A. M. Nicolson,[32] who used Rochelle salt as the active material. Rochelle salt is unique in that its piezoelectric constant is about a thousand times as great as that of any other crystal. It has, however, several characteristics which would appear to render it unsuitable for use as a measuring microphone. It is mechanically fragile and its piezoelectric activity, under normal conditions, varies greatly with temperature, falling to a very low value for temperatures above 23° C. R. D. Schulwas-Sorokin [33] found, however, that by the application of a static stress the temperature coefficient could under certain conditions be greatly reduced and the activity extended to higher temperatures. C. B. Sawyer [34] found that if two thin slabs are cut and cemented together in such a way that one of the slabs will expand and the other contract when a potential is applied between the interface and the two outer surfaces, variations of activity with temperature are reduced to a low value. Presumably stresses are set up in the slabs by temperature variations which reduce the temperature coefficient of activity in accordance with the experiments of R. D. Schulwas-Sorokin. Sawyer has utilized these so-called bimorphic slabs in the construction of microphones. Single elements can be constructed of sufficiently small dimensions to avoid diffraction of the sound. In order to obtain microphones of greater practical efficiency a number of elements may be used in combination. If these elements are mounted symmetrically the translating efficiency will be the same in all directions about the axis of symmetry, as is the case for any microphone having an axis of symmetry. The amount of variation in respect to other directions depends upon the relation between the dimensions and the wave-length. According to the published data the sensitivity of a multiple element microphone of this type is about 25 db below that of a moving coil instrument.[35]

[31] Gaines, *Physics* 3, 209 (1932).
[32] *Trans. A. I. E. E.* 38, 1315 (1919).
[33] *Zs. f. Physik* 73, 9–10, 700 (1932).
[34] *Proc. I. R. E.* 19, 2020 (1931).
[35] A. L. Williams, *Jour. S. M. P. E.* 23, 196 (1934).

Of the more common piezoelectric crystals tourmaline possesses a characteristic which renders it peculiarly suitable for the absolute measurement of sound intensities at all audio frequencies, in that it may be so cut into slabs that a potential difference will be developed between its lateral surfaces when it is subjected to a purely hydrostatic pressure. Because of this characteristic Sir J. J. Thomson [36] suggested its use for measuring pressures in gun barrels. Such a slab of tourmaline, having dimensions small compared with the wavelength, except for its low sensitivity, is the ideal microphone. Tourmaline is mechanically strong and its activity is practically constant under all atmospheric conditions. Resonant frequencies in the slab lie far out of the range of audio frequencies so that the response at all frequencies is the same and is easily determined from static or low-frequency measurements. Unfortunately the sensitivity of such a device is low, some 70 db below that of a moving coil microphone. In spite of this low sensitivity it can be used for calibrating other microphones if sound waves of rather high intensities are used and if the electrical circuit is provided with a band-pass element transmitting only frequencies in the immediate neighborhood of the measuring frequency. A measuring system of this character, unlike the Rayleigh disc, is not subject to disturbances from circulating air currents.

Thermometric Microphones

As the pressure variations in a sound wave are accompanied by corresponding variations in temperature, corresponding electrical currents will be generated by a resistance thermometer or a thermocouple when placed in the sound field. The temperature variations are of the order of 0.0001° C. per bar acoustic pressure. The use of a resistance thermometer (for measuring these periodic temperature variations) was first investigated by Heindlhofer,[37] and more recently by Friese and Waetzmann,[38] who found that at a frequency of 1000 c.p.s. a wire 0.0004 cm. in diameter will undergo temperature variations equal to about 0.15 of the variations in the surrounding medium. To derive an alternating electric current from the periodic resistance variations that follow the temperature variations, a direct current must be passed through the wire. The heat generated by this current, unless it is kept down to an extremely small value, will set up convection currents around the wire and so greatly complicate the operation.

The thermocouple is entirely free from this objection, but is not readily constructed so as to have a heat capacity as small as the Wollaston wire. Recently A. E. Johnson [39] has been able to make thermocouples with exceedingly small heat capacities with which measurements have been made up to 5,000 cycles and it is stated that they are usable up to several hundred thousand cycles. They are so small that they do not alter the sound field by diffraction and are free from resonance effects inherent in most instruments depending upon mechanical movement. As compared with other types, thermocouple microphones have a low sensitivity, at least 100 db below that of the moving coil microphone, according to the data given by Johnson.

[36] *Engineering* 107, 543 (1919).
[37] *Ann. d. Physik* 37, 247 (1912); 45, 259 (1914).
[38] *Zs. f. Physik* 29, 110 (1925); 31, 50 (1925); 34, 131 (1925).

Velocity Microphones

All the preceding types of microphones depend ultimately for their operation upon pressure variations in the sound wave. As the two primary characteristics of sound are pressure variations and alternating flow of the air particles, it is possible also to design microphones which generate voltages in accordance with the velocity of the air particles.

Hot Wire Microphone

One form of microphone of this character depends upon the change in resistance of a heated fine wire resulting from changes in temperature produced by the transverse flow of air. A microphone operating on this principle was first devised by Tucker [40] and used extensively during the war for locating enemy artillery. In order to increase the sensitivity and reduce distortion a steady stream of gas should be passed across the wire. An application of this principle to the construction of a microphone is shown in Fig. 4. Maximum response is

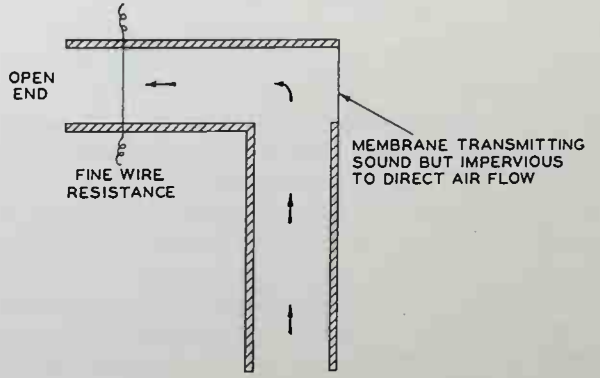

Fig. 4—Hot wire velocity microphone.

[39] *Phys. Rev.* 45, 645 (1934).
[40] *Phil. Trans.* 221, 389 (1921).

obtained when the direction of the sound wave coincides with the direction of the steady stream. At a given frequency the resistance variation is nearly proportional to the product of the steady stream velocity, the particle velocity of the sound wave and the cosine of the included angle. Since velocity in contrast with pressure is a vector quantity, a velocity microphone will respond selectively to sound coming from certain directions even at low frequencies. This characteristic is of considerable advantage in certain types of measurements, for it is often possible to so place and orient the instrument that its response is a minimum for an interfering or disturbing sound and a maximum for the sound to be measured. An illustration of such an application in sound measurement or pick-up is shown in Fig. 5

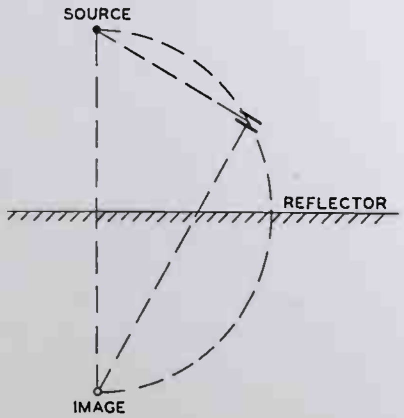

Fig. 5—Method of reducing effect of interfering waves with velocity microphone.

where the instrument is so placed that it will receive the sound directly from the source, but is insensitive to the sound reflected from the floor or neighboring wall. Other examples of acoustic measurements, where benefit is derived from the directional characteristics of the velocity microphone, are discussed in a recent paper by Wolff and Massa.[41]

There is one other important difference in the performance of velocity and pressure microphones. In a plane progressive wave particle velocity and pressure are strictly proportional at all frequencies. For a spherical sound wave, the radius of curvature of which is small compared with a wave-length, this is no longer true.

[41] *Jour. Acous. Soc. Amer.* IV, 217 (1933).

If we have a simple source of constant strength $A \cos kct$, the pressure at a distance r is given by

$$\frac{A\rho f}{2r} \sin k(ct - r),$$

where ρ is the density, c the velocity of sound, and k is equal to ω/c. The particle velocity is given by

$$\frac{Af}{2cr}\left[1 + \left(\frac{\lambda}{2\pi r}\right)^2\right]^{1/2} \sin\left[k(ct - r) + \psi\right],$$

where ψ is a function of λ and r. It will be seen that the pressure varies inversely with the distance for all frequencies, while the relationship between velocity and distance involves the wave-length, or frequency. In Fig. 6 are given some response-frequency characteristics

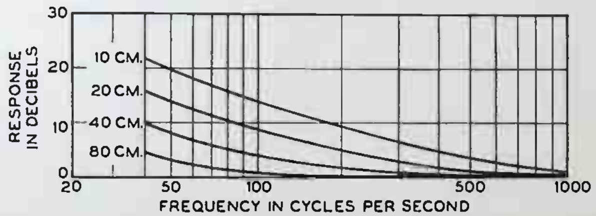

Fig. 6—Response of velocity microphone as a function of distance from source.

for several distances from the source of a velocity microphone having a uniform characteristic for plane waves. The change in characteristic as the instrument is brought close to the source is very marked. Some care is therefore required in interpreting the results of measurements made with this type of instrument. On the other hand, a pressure microphone, except in so far as diffraction may modify the sound field, will exhibit the same form of response-frequency characteristic at all distances from the source, so the wave form of the voltage generated by a pressure microphone will be the same for all positions in the free sound field of a simple source. This difference in characteristics of the two types of instruments is easily observed by comparing reproduced speech when the microphones are first placed near and then at some distance from the speaker's mouth.

Ribbon Microphone

A form of microphone, which has been extensively used in recent years, is the ribbon microphone. Essentially it consists of a very thin strip of aluminum with circuit terminals at its two ends. This ribbon is placed in a magnetic field so that the lines of force lie in the plane of the ribbon and perpendicular to its long dimension, as shown in Fig. 7. Motion of the ribbon set up by sound waves will then generate a potential between its terminals. This type of microphone construction was first suggested by Reinganum.[42] It was developed into a practical form by Gerlach [43] and Schottky.[44] They apparently preferred to shield one side of the ribbon so that the instrument operated as a pressure microphone. H. F. Olson,[45] recognized the greater simplicity of the instrument in construction and in operation if both sides of the ribbon were freely exposed to the air. Constructed in this

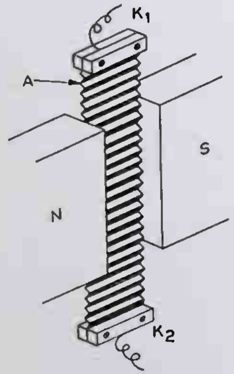

Fig. 7—Ribbon microphone.

way the instrument is virtually a velocity microphone at least at low frequencies. At the higher frequencies the ribbon with its surrounding structure almost completely shields the rear from sound reaching the front of the ribbon at perpendicular incidence. Under these conditions the instrument operates substantially as a pressure microphone, but even at these frequencies sound reaching the instrument from a direction parallel to the plane of the ribbon is without effect. Curves published by Olson on the directional characteristics of this microphone show that in a plane perpendicular to the axis of the ribbon the variation of response with direction follows approximately a cosine law, where the angle is measured from a line drawn normal to the plane of the ribbon. The relationship is more complicated over a plane passing through the axis of the ribbon.

[42] *Phys. Zs.* 11, 460 (1910).
[43] *Phys. Zs.* 25, 675 (1924); *Wiss. Ver. Siemens-Konz* 3, 139 (1923).
[44] *Phys. Zs.* 25, 672 (1924).
[45] *Jour. Acous. Soc. Amer.* 3, 56 (1931).

A microphone that responds to the velocity of the air particles has equal sensitivity for sound waves traveling in opposite directions. Weinberger, Olson and Massa [46] have modified the ribbon microphone so that its response is not the same for the positive as for the negative direction of propagation of the sound wave. So modified the microphone has a greater response for sound waves coming towards one side of the ribbon than for those coming towards the other side. In a plane wave the magnitude of the pressure and the velocity are proportional. For waves traveling in a positive direction the two are in phase and for waves traveling in the negative direction they are in opposite phase. If, then, a pressure and a velocity microphone of equal sensitivity are connected in series the resultant voltage will be double for sound of normal incidence coming from one direction and equal to zero for sound coming from the opposite direction. Weinberger, Olson and Massa placed an appropriate acoustic impedance over a part of the ribbon so as to give this part the characteristics of a pressure microphone, while the other part of the ribbon was left free so as to function as a velocity microphone. The voltage at the ends of the ribbon is then proportional to the vector sum of the pressure and the velocity in the sound wave. In this way a pressure and velocity microphone combination is obtained in one instrument. It is insensitive to sound falling at perpendicular incidence on one side of the ribbon but not to sound propagated in the plane of the ribbon, as in the case of a velocity ribbon microphone.

Electrical Instruments of Particular Interest In Acoustical Studies

So far our discussion has been restricted mainly to microphones and instruments used in their calibration. There have, of course, in recent years been developed many other devices especially adapted for the quantitative study of particular acoustic problems, but a rather extensive discussion of the microphone has been given because it is an adjunct in almost all of these other instruments. In great part acoustic measurements are today made by first translating sound into a corresponding amplified electric current. The results of measurement or analysis of this current may then be referred back to the sound if the characteristics of the translating device are known. The type of analyzer or measuring instrument applied to the electrical circuit depends then altogether upon the kind of information that is desired. Strictly speaking we should classify these not as acoustical but as electrical instruments. In fact, every kind of electrical instru-

[46] *Jour. Acous. Soc. Amer.* 5, 139 (1934).

ment may at times find an application in the study of acoustical problems. We must, therefore, necessarily restrict ourselves to a discussion of the kind of instruments which can give types of information of general acoustical interest.

The Oscillograph

If we wish to obtain a complete picture of the sound wave the microphone amplifier output is connected to an oscillograph, which is a device for translating a time pattern of the electric current into a corresponding space pattern. If an undistorted pattern is to be obtained, not only must the various harmonic components of the current and the recorded wave have the same relative amplitudes, but their phase relationships must be preserved. One form of instrument very closely satisfying these conditions up to about 10,000 cycles is a Curtis string oscillograph,[47] which is a modified form of the Einthoven galvanometer. The arrangement of this instrument is shown diagrammatically in Fig. 8. It records the wave form optically on photographic paper, which is automatically developed and fixed within a fraction of a minute after exposure.

In certain types of problems one of the various recording devices employed in the production of sound pictures may be used advantageously. These instruments have, in general, not been designed to be free from phase distortion but the records are in a form suitable for reproduction so they lend themselves particularly to the study of the subjective aspects of sound.

When the sound wave to be studied is steady, the wave form can be conveniently observed or photographed by means of a cathode ray oscillograph. These instruments are now to be had in convenient form. When used with an automatic sweep circuit, as suggested by Bedell and Reich,[48] the wave form of any steady state current is shown as a stationary pattern on a screen. These oscillographs are generally free from both frequency and phase distortion up to the highest audio frequencies.

Harmonic Analyzers for Steady Currents

If we wish to study the composition of a sound wave in terms of its harmonic components we may, of course, analyze the oscillographic records by means of any one of the well known methods of harmonic analysis, but this is at best a laborious process. Also, it is usually difficult to read an oscillogram with sufficient accuracy to determine the magnitude of any component that is much smaller than that of the maximum component. When the conditions are such that a current can be steadily maintained, the analysis can be made much more conveniently and generally over a wider range of amplitudes by means of one of a number of recently developed types of current analyzers. These analyzers, although the mode of operation may differ in the various types that are available, have the common characteristic that they transmit only a narrow band of frequencies with the position of the band adjustable along the frequency scale. For analyzing a current wave the mid-frequency of the transmission band is shifted along the whole frequency range and the magnitude of the current in each frequency region is recorded or noted on a meter. This type of analyzer may also be used to get a statistical distribution of the power with respect to frequency when the sound is not periodic, as in certain kinds of noise.

High-Speed Analyzers

For the rather detailed study of sounds whose wave form varies with time, such as those of music or speech, an instrument is needed which

[47] *Bell Sys. Tech. Jour.* XII, p. 76.
[48] *Science* 63, 619 (1926).

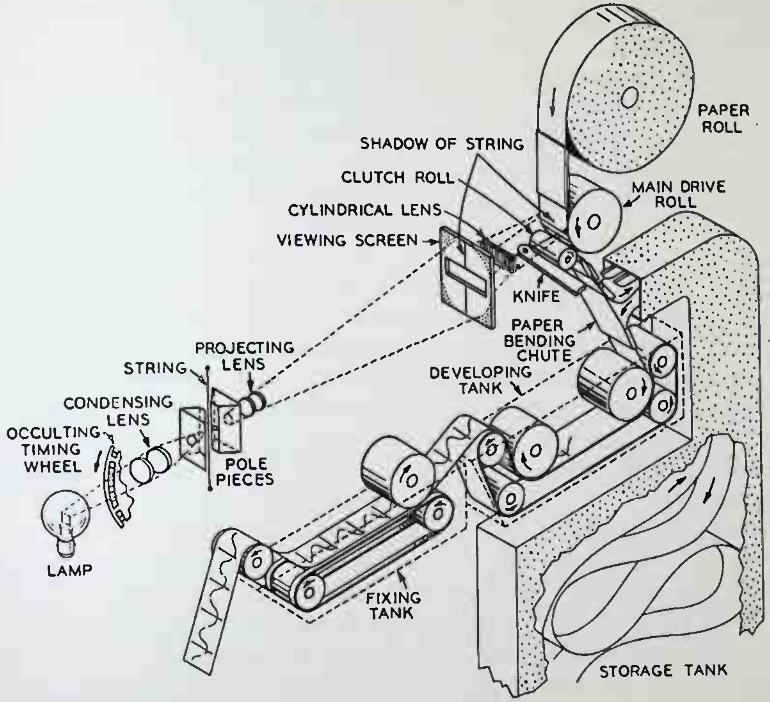

Fig. 8—Diagrammatic arrangement of Curtis oscillograph.

shall indicate the variations with time in the frequencies and amplitudes of all the harmonic components. Analyzers of the type just described indicate at a given instant the amplitude of only one component. In order to follow variations in all the components it would be necessary to sweep the frequency of transmission of the analyzing circuits rapidly back and forth over the frequency range of interest. The selective element of the analyzer, however, possesses a finite time constant; that is, when the selective circuit is set at a given transmission frequency a finite time is required for the transmitted current to reach a certain fraction of its steady state value and, similarly, a finite time is required for the current to decay to a certain fraction of this value when the transmission frequency is changed. This time constant depends to some extent upon the shape of the transmission versus frequency characteristic of the analyzing circuit but, in general, it bears an inverse relation to the selectivity. It is therefore not possible with an analyzer of this type, having a single variable selective element, to perform a rapid analysis without sacrificing resolution. This difficulty can, however, be circumvented if the analyzer is provided with a large number of fixed selective elements which are continuously operative. To build up the large number of required circuits from electrical elements would be extremely costly and would result in a bulky piece of apparatus. A compact form of analyzer having a large number of fixed selective mechanical elements has recently been described by C. N. Hickman.[49] This device has a series of tuned reeds, all driven electromagnetically at the same time by the current to be analyzed. The reeds are tuned so that their resonant frequencies differ progressively by equal pitch intervals. One hundred and twenty reeds are used to cover the range from 50 to 3,200 cycles. The deflection of each reed is made visible by the projection on a screen of a spot of light reflected from a mirror attached to the reed. The strength of each component in the current may thus be observed simultaneously on the screen or, if desired, the deflections may be recorded photographically.

A different and ingenious approach to this problem has been made by E. Meyer[50] in a recently described instrument. By methods well known in communications engineering the frequency of each component in the current to be analyzed is increased by an equal amount. A special high-frequency loud speaker translates the resultant currents into sound waves which are now all of very short wave-length. These waves are reflected from a concave grating made up of a large number

[49] *Jour. Acous. Soc. Amer.* 6, 108 (1934).
[50] *Zeits. für Tech. Phys.* 12, 630 (1934).

of equally spaced rods. The component waves are brought to a focus at different points along a focal surface analogous to the dispersion of light waves by an optical grating. A high-frequency microphone is moved back and forth along the focal plane through an amplitude large enough to cover one order of the spectra. This microphone is connected to an appropriate meter which records optically the intensity at various parts of the spectrum, which have a 1:1 correspondence with the component frequencies in the original current.

Measurement of Pitch

For acoustical studies, where it is of no particular importance to know the wave form but where interest lies in the variation of pitch with time, as in the study of the vibrato in musical tones, or in the inflections of the speaking voice, several types of instruments have been devised. Perhaps of these the most widely known is the tonoscope developed by C. E. Seashore[51] and his associates, which operates on the stroboscopic principle. This instrument has rows of uniformly spaced dots on a rotating cylinder, the number of dots increasing in successive rows. A neon light is made to flicker in synchronism with the fundamental of the tone under investigation. The particular row which under the light appears stationary gives the pitch of the tone at any instant. By the aid of a suitable camera the time variations of pitch may be recorded photographically, giving a so-called strobophotograph.

A frequency recorder operating on a different principle has been described by Hunt.[52] By a special circuit arrangement, employing gas-filled discharge tubes in combination with a spark recorder, the pitch of a tone can be recorded on paper. The scale is linear up to 8,000 cycles. This instrument is capable of following changes in pitch at a high rate of speed.

High-Speed Level Recorder

In some important types of sound measurements we are not interested in a detailed analysis of the sound wave but merely in the variation with time of the average level of the sound, as in the measurement of the rate of decay in a room or the flow of energy in speech, music, or noise. In some cases this average is preferably taken over long and in others over short time intervals. For long time averages, a thermocouple or rectifier and an ammeter may be used, but for short time averages an instrument is required which can follow changes in intensity at a higher rate of speed. Frequently also the range of inten-

[51] *Jour. Acous. Soc. Amer.* 2, 77 (1930).
[52] *Rev. Sci. Inst.* 6, 43 (1935).

sities over which we desire to make measurements of this character is very wide. Reverberation measurements are preferably made over a range of at least 60 db and the level range of orchestral music covers about 75 db. Several instruments designed for such purposes have been described recently.[53] In the instrument described by Wente, Bedell and Swartzel the level is recorded by a stylus on waxed paper. The recorder can be adjusted to give either a short or a long time average. At the higher speeds it is capable of following changes in intensity at the rate of 840 db per second and fluctuations in intensity of about 100 per second. The instrument may be adjusted so that the full scale covers a range of 30, 60 or 90 db.

Loudness Measurements

The preceding discussion was restricted to the purely objective or physical aspects of sound. In certain types of acoustical problems, as in the study of noise, we are, however, interested in subjective characteristics, but we do not yet have instruments which respond to an acoustic stimulus in the way the brain does through the ear. In fact

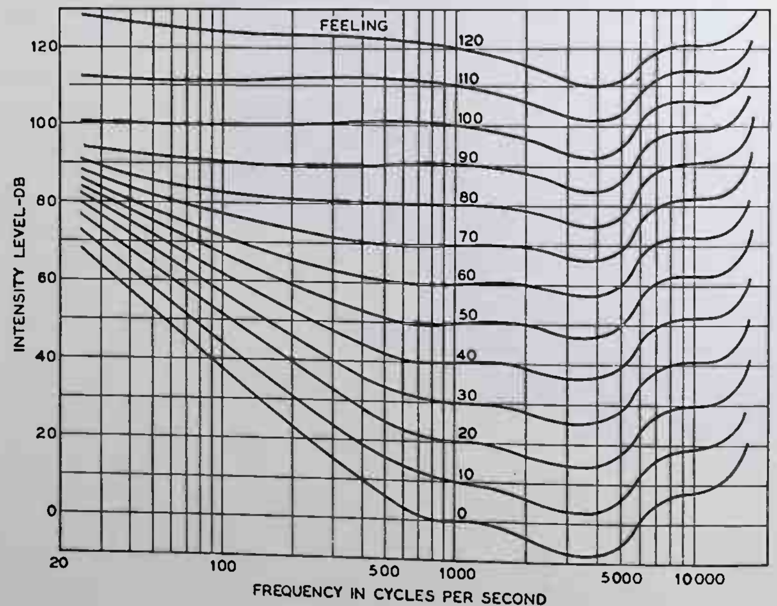

Fig. 9—Auditory chart.

[53] Hunt, *Jour. Acous. Soc. Amer.* 6, 54 (1934). Wente, Bedell and Swartzel, *Jour Acous. Soc. Amer.*, January, 1935.

we are not yet completely clear as to the relationship between sensation and stimulus, although Fletcher and Munson[54] have developed formulae whereby the loudness level of a steady sound can in most cases be computed from the intensity level of its components. For single pure tones the relationship between sensation and stimulus has been extensively explored with the results which are indicated in the auditory chart shown in Fig. 9, as given by Fletcher and Munson. The various curves give the intensity level of pure tones of equal loudness. This chart gives some idea of the complexity of the relationship between loudness and stimulus. The threshold of audibility of course varies widely with frequency and the relationship between sensation level and intensity level is not the same at the various frequencies and levels; for instance, at a loudness level of 40 db above threshold, a change of 5 db in the stimulus at 100 cycles produces the same change in sensation as a change of 10 db at 1,000 cycles. Fechner's law does not hold strictly over a wide range of intensities at any of the audible frequencies. The difficulty of devising an instrument which would have similar characteristics is apparent. "Sound level meters" have, however, been designed which give a reading which is approximately proportional to the subjective intensity of the sound. These meters are generally so designed that they have a frequency characteristic corresponding to the auditory curve at about the level of the noise being measured. They have proved themselves extremely useful although from our knowledge of hearing phenomena we might expect large variations in the actual loudness of sounds of different character, even if a noise meter of the above type should show them to be equal.

[54] *Jour. Acous. Soc. Am.* 5, 82 (1933).

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JULY 1930

MONOGRAPH B-500

A CHRONOGRAPHIC METHOD OF MEASURING REVERBERATION TIME

BY

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A CHRONOGRAPHIC METHOD OF MEASURING REVERBERATION TIME

BY E. C. WENTE AND E. H. BEDELL
Bell Telephone Laboratories

Reverberation time measurements are generally made with the ear and a stop watch in the manner devised by Prof. Wallace Sabine. Surprisingly consistent results can be obtained by this method in a reverberation chamber, where the rate of decay of sound is slow and where disturbing sounds are absent. But such measurements present difficulties if the room is noisy or if the reverberation time is short. Also it is recognized that uncertainties may be introduced because of the fact that the threshold of hearing varies between individuals and with time in the same person. It was with the object of overcoming these difficulties that the electrical method to be described was devised. This method does not differ essentially from that of Sabine except that an

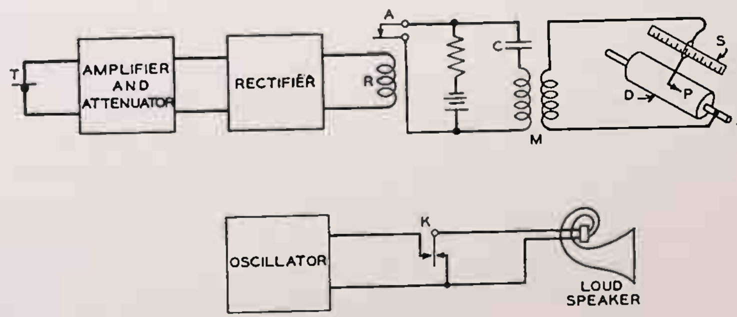

FIG. 1. *Schematic of Apparatus.*

electro-acoustical ear of controllable threshold sensibility is substituted for the human ear.

The general principles of construction of this mechanical ear are indicated in Fig. 1. T is a microphone connected to a vacuum tube amplifier provided with an attenuator with which the gain can be controlled in definite logarithmic steps. The amplifier is terminated by a double wave rectifier. The rectified current passes through the receiving winding of a relay, R. This relay is so constructed that when the rectified current exceeds a certain value the armature opens the contact at A. With this contact open the condenser, C, is charged from the battery, E. When the current falls below a certain value, the armature is released, contact is made at A, and the condenser discharges through the primary winding of the spark coil, M, causing a spark to pass to a rotating drum, D, at P. This spark circuit is similar to that described by C. N. Hickman.[1] The amplifier, rectifier and relay may be so designed that whenever the sound intensity actuating the microphone decreases to a certain value a spark will pass to the drum. This value of intensity may be designated the threshold sensibility of the apparatus.

The drum, D, is rotated at a known, constant speed by an electric or a spring motor. The discharge point may be set at any place along the axis of the drum, the position being indicated on the scale, S. A sheet of waxed paper is placed about the cylindrical surface of the drum, on which a spark discharge leaves a permanent impression.

In the measurement of reverberation time the source of sound may be the usual organ pipe, although a loud speaker driven from an oscillator has generally been used with this apparatus. The tone from the loud speaker can be interrupted by the key, k, which opens the circuit from the oscillator and at almost the same instant short-circuits the speaker. This key is provided with a trigger, not shown, which, after having been set, is released mechanically at a definite angular position of the drum. The time required for the drum to rotate from this position to its position when the spark passes is the time required for the sound to die down to the threshold sensibility of the apparatus.

Measurements of reverbation time are carried out in the following manner. The loud speaker and microphone are placed in the room to be measured. The attenuator is adjusted to a certain value and the spark point is set at some definite position on the scale, S. After the sound has reached a steady state, the trigger of the key, k, is set. This is subsequently released by the rotating drum and the sound from the loud speaker is thereby interrupted. An interval of time later the spark jumps to the drum, leaving an impression on the paper. The gain of the amplifier is then changed by a definite number of *db* and the spark point moved along the scale, S, a corresponding distance; e.g., if the gain of the amplifier is increased by 3 *db* the position of P is moved to the right a half inch and for 6 *db*, one inch. Similarly if the gain is decreased by 3*db*, P is moved a half inch to the left. In this way a series of dots covering a wide attenuation range can be obtained.

[1] Journal of the Acoustical Society, Vol. 1, No. 1, p. 138.

The paper when removed from the drum should then show a succession of dots similar to that shown in Fig. 2. If the decay of the sound at the microphone had been strictly logarithmic these dots would all lie along a straight line. This ideal condition will almost never be encountered in practice. We must therefore be content with drawing a line of best fit through these points. From the slope of the line so drawn and the peripheral speed of the drum, the reverberation time as usually defined is readily obtained.

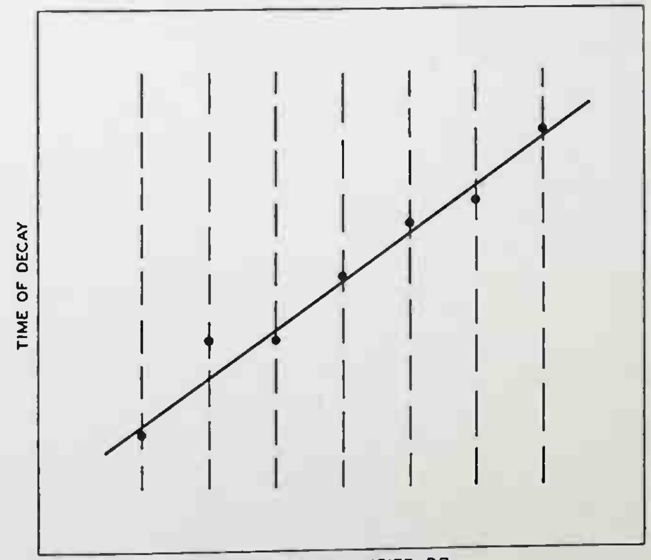

Fig. 2

Due to the shift of the interference pattern as sound dies out in a room the pressure on the microphone will not be a continually decreasing function of the time. In fact, in most cases the energy delivered by the microphone will pass in rapid succession through any previously selected value several times before finally remaining below that value. With the apparatus as described we would get then a group of several sparks as the average energy level passed through the threshold value. Some method must therefore be devised for averaging out the fluctuations in the energy decay curve. The method we have adopted is that of inserting an electrical filter consisting of a single capacitance and inductance between the rectifier tubes and the relay winding. This filter introduces some time lag in the measurements, the lag being a function of the size of the elements of the filter and of the rectified current through the filter. The inductance and capacitance of the filter should therefore be made as small as is consistent with insuring a single spark for each time interval. The electrical damping in the filter may be made large compared with the acoustical damping in most rooms so that the use of the filter does not affect the measurements; but in order to insure the validity of our results in very dead rooms we have designed the amplifier and re-

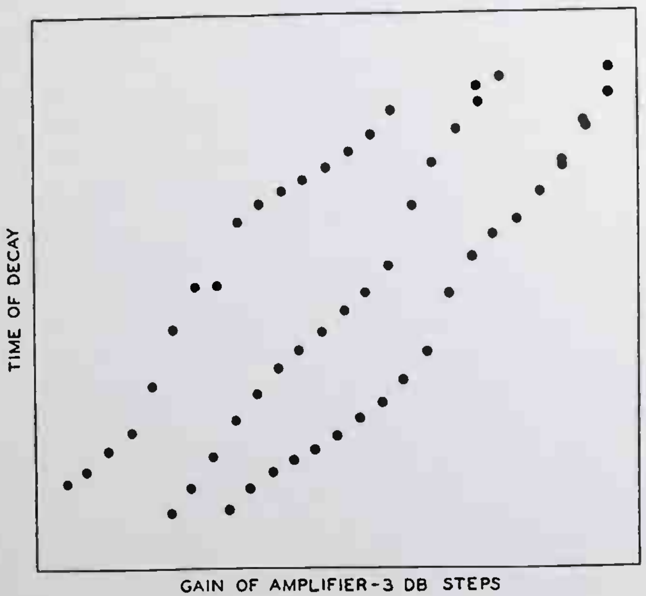

FIG. 3

ctifier so that the rectifier overloads at a level just comfortably above that necessary to operate the relay. This gives practically a constant current through the filter and relay independent of the amplifier gain, or of the output of the source, over the range of values used. The time lag for each interval measured is very nearly constant therefore, so that the slope of the line obtained is not affected by the delay in the filter.

An indication of the reliability of the measurements may be gained by a comparison of the records obtained with the microphone placed at various positions in the room, other conditions being kept the same.

Fig. 3 shows three such records. Although these records differ considerably from each other, yet, when a straight line is drawn through each set of points, it will be found that the slopes of these lines are nearly the same. These records show, however, the desirability of making measurements over a wide intensity range.

The wavy character of the succession of dots may be much reduced if instead of a pure tone a frequency modulated tone is used.[2] Fig. 4 shows two series of points, the upper one of which was obtained with a pure tone and the lower with a tone having its frequency varied sinu-

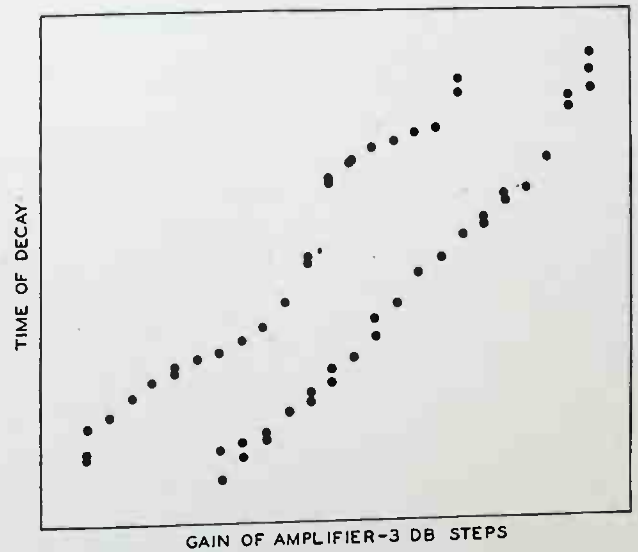

FIG. 4

soidally seven times per second from 1025 to 975 c.p.s. It is seen that in the latter case the points lie much closer along a straight line.

It has also been found a help to place over the microphone a Helmholtz resonator tuned to the measuring frequency. This expedient is particularly helpful in noisy rooms. But even where there is no noise it permits measurements to be made over a wider intensity range, the limit in this case being set by amplifier noise. The time constant of the resonator is, in general, short compared with that of the room, so that no appreciable error is introduced by the use of a resonator, al-

[2] E. Meyer and P. Just, Elektr. Nachr-Techn. 5, 293 (1928).

though at low frequencies in highly damped rooms the damping constant of the resonator may have to be taken into account.

In some of our measurements we have used two microphones with amplifiers and rectifiers of identical characteristics connected in parallel to the relay winding. The microphones were in this case placed in different parts of the room. With this arrangement the dots obtained lie more nearly along a straight line. But except in the case of permanent installations the extra complication introduced can probably not justify itself.

It may be mentioned that instead of varying the gain of the amplifier we may change the input power of the loud speaker and practically the same results will be obtained, unless the loud speaker has a non-linear input-output characteristic.

For successful operation of this apparatus it is necessary that there be no variation in the characteristics of the amplifier and the relay only during the time that one set of points is taken. This condition can be satisfied without difficulty. The amplifier must have a high gain in order to cover a wide range of intensities, but it is not necessary that it be of large power capacity, as no errors will be introduced if the amplifier is overloaded at levels greater than that necessary to operate the relay.

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HEADQUARTERS

The headquarters of the Convention will be the Sagamore Hotel, where excellent accommodations and moderate rates are assured. A reception parlor will be provided as headquarters for the Ladies' Committee.

Hotel reservation cards mailed to the members of the Society several weeks ago should be filled out and mailed immediately to the Sagamore Hotel so that suitable accommodations may be reserved, subject to cancellation if unable to attend the convention.

The following European-plan day rates are extended by the Sagamore Hotel to Society members and guests attending the Convention (all rooms are outside rooms with bath):

Room for one person	\$3.00 to \$5.00
Room for two persons, double bed	4.50 to 6.00
Room for two persons, twin beds	6.00 to 7.00
Suite accommodations, one to four persons	12.00 and up

The following hotel garage rates will be available to SMPE delegates and guests who motor to the Convention: 24-hr. inside parking, 75¢; outside parking (daily), 25¢.

The colorful *Sagamore Room* on the main floor of the Hotel offers special breakfast, luncheon, and dinner menus at moderate prices.

Golfing privileges at several Rochester country clubs may be arranged for either by the hotel management or at the SMPE registration headquarters.

Tentative Program

MONDAY, MAY 5, 1941

9:00 a.m. *Glass House (Hotel Roof)*; Registration.

9:30 a.m. *Glass House (Hotel Roof)*; General Session.

Report of the Convention Arrangements Committee; W. C. Kunzmann, *Convention Vice-President.*

Report of the Financial Vice-President; A. S. Dickinson.

Report of the Engineering Vice-President; D. E. Hyndman.

Welcome by the President; Emery Huse.

"The University of Minnesota Visual Education Program"; R. A. Kissack (*Demonstration*).

"Five New Models of 16-Mm Sound Kodascopes"; W. E. Merriman, Eastman Kodak Co., Rochester, N. Y.

"Aviation Films—A New Frontier for Motion Picture Production"; Arch A. Mercey, Office of Government Reports, Executive Office of the President; and E. M. Ellingson, Civil Aeronautics Board, Washington, D. C.

"Air-Conditioning Safety Device for Motion Picture Theaters"; E. R. Morin, Motion Picture Division, Connecticut State Police.

"The Specialization of Film Delivery"; J. H. Vickers, National Film Carriers, Inc., Philadelphia, Pa.

12:30 p.m. *Starlight Room;* Informal Get-Together Luncheon; President Huse, *Chairman.*

Addresses by:

The Honorable S. B. Dicker, Mayor of the City of Rochester.

Dr. F. R. Watson, President, Acoustical Society of America.

Dr. Howard Hanson, *Director*, Eastman School of Music, University of Rochester, Rochester, N. Y.

2:00 p.m. *Glass House (Hotel Roof)*; Illumination Session.

"Characteristics of Intermittent Carbon Arcs"; H. G. MacPherson, R. B. Dull, and F. T. Bowditch, National Carbon Company, Cleveland, Ohio.

"Some Properties of Polished Glass Surfaces"; F. Jones, Bausch & Lomb Optical Co., Rochester, N. Y.

"Improvements in Methods of Surface Treatment of Lenses"; W. Miller, Vard Mechanical Laboratories, Pasadena, Calif.

Report of the Standards Committee; D. B. Joy, *Chairman.*

"A Method for Designing Film Sprockets"; W. G. Hill and C. L. Schaeffer, Agfa Ansco Corp., Binghamton, N. Y.

Report of the Non-Theatrical Equipment Committee; J. A. Maurer, *Chairman.*

"Report on the Activities of the Inter-Society Color Council"; R. M. Evans, *Chairman* of SMPE Delegates.

8:00 p.m. *Glass House (Hotel Roof)*; Sound Session.

"Informal Report on Control-Tracks and Multiple Horns, from the Academy Research Council"; J. O. Aalberg, Hollywood, Calif.

"Multi-Speaker Systems"; H. I. Reiskind, RCA Manufacturing Co., Inc., Indianapolis, Ind.

"Fantasound"; N. A. Hawkins and W. Garity, Walt Disney Studios, Burbank, Calif.

"Vitasound"; N. Levinson and L. T. Goldsmith, Warner Bros.-First National Studios, Burbank, Calif.

"Motion Picture Technic and Multi-Horn Reproduction"; L. L. Ryder, Paramount Pictures, Inc., Hollywood, California

"New and Old Aspects of the Origins of 96-Cycle Distortion"; J. O. Baker and R. O. Drew, RCA Manufacturing Co., Inc., Camden, N. J.

TUESDAY, MAY 6, 1941

9:00 a.m. *Glass House (Hotel Roof)*; Registration.

9:30 a.m. *Glass House (Hotel Roof)*; General and Business Session.

"New Gadgets for the Film Laboratory"; B. Robinson and M. Leshing, Twentieth Century-Fox Film Studios, Hollywood, Calif.

"An All-Purpose Sound-Track Printer"; G. M. Best, Warner Bros.-First National Studios, Burbank, Calif.

"Some Equipment Problems of the Direct 16-Mm Producer"; L. Thompson, The Calvin Co., Kansas City, Mo.

11:00 a.m. Society Business.

"Some Recent Advances in the Theory of the Photographic Process"; C. E. K. Mees, Eastman Kodak Co., Rochester, N. Y.

"Performance of the Visual Mechanism in the Viewing of Motion Pictures"; Brian O'Brien, Institute of Applied Optics, University of Rochester, Rochester, N. Y.

2:00 p.m. Open Afternoon.

7:30 p.m. *Starlight Room;* 49th Semi-Annual Banquet and Dance.

Dancing and Entertainment.

WEDNESDAY, MAY 7, 1941

Joint Meeting of the Acoustical Society of America and the Society of Motion Picture Engineers

9:30 a.m. *Eastman Theater;* Symposium on the Stereophonic Sound-Film System, by the Bell Telephone Laboratories, New York, N. Y.

"General Theory"; Harvey Fletcher and E. C. Wente.
"Mechanical and Optical Equipment"; E. C. Wente, R. Biddulph, L. A. Elmer, and A. B. Anderson.
"Theory and Performance of Compandor Systems"; Harvey Fletcher and W. B. Snow.
"Pre- and Post-Equalization of Compandor Systems"; J. C. Steinberg.
Technical Demonstration.

Joint Meeting of the Acoustical Society of America and the Society of Motion Picture Engineers

2:00 p.m. *Glass House (Hotel Roof);* General Session.

"Electrical Equipment for the Stereophonic Sound-Film System"; W. B. Snow and A. R. Soffel.
"A Light-Valve for the Stereophonic Sound-Film System"; E. C. Wente and R. Biddulph.
"Internally Damped Rollers"; E. C. Wente and A. H. Müller.
"A Non-Cinching Film-Rewind Machine"; L. A. Elmer.

"Progress in Three-Dimensional Photography"; J. A. Norling, Loucks & Norling, New York, N. Y. *(Demonstration.)*
"Solar Prominences in Motion"; R. R. McMath, McMath-Hulbert Observatory, Lake Angelus, Pontiac, Mich. *(Demonstration.)*

Joint Meeting of the Acoustical Society of America and the Society of Motion Picture Engineers

8:00 p.m. *Eastman Theater;* Demonstration of Stereophonic Sound-Film System, by the Bell Telephone Laboratories, New York, N. Y.

THURSDAY, MAY 8, 1941

9:30 a.m. *Glass House (Hotel Roof);* Projection Session.

Report of the Theater Engineering Committee; A. N. Goldsmith, *Chairman.*

Sub-Committee on Projection Practice; H. Rubin, *Chairman.*
Sub-Committee on Theater Design; B. Schlanger, *Chairman.*
Sub-Committee on Screen Brightness; F. E. Carlson, *Chairman.*

Projection Practice Symposium, prepared for the Projection Practice Sub-Committee of the Theater Engineering Committee.

"Projection Room Equipment Requirements"; J. J. Sefing.
"The Projection Room—Its Location and Its Contents"; J. R. Prater.
"Factors Affecting Sound Quality"; A. Goodman.
"Factors to Be Considered in a Sound Screen"; C. F. Holly.

"A Suggested Clarification of Carbon Arc Terminology as Applied to the Motion Picture Industry"; H. G. MacPherson, National Carbon Co., Cleveland, Ohio.
"Improved Methods of Controlling Carbon Arc Position"; D. J. Zaffarano, W. W. Lozier, and D. B. Joy, National Carbon Co., Fostoria, Ohio.
"A New 13.6-Mm High-Intensity Projector Carbon"; W. W. Lozier and D. B. Joy, National Carbon Co., Fostoria, Ohio.

2:00 p.m. *Glass House (Hotel Roof);* General Session.

"The Subjective Sharpness of Simulated Televison Images"; M. W. Baldwin, Jr., Bell Telephone Laboratories, New York, N. Y.
"A Compact Direct-Reading Reverberation Meter"; E. S. Seeley, Altec Service Corp., New York, N. Y.
"Solution of Acoustic Problems Encountered in Recording Sound for Motion Pictures"; W. L. Thayer, Paramount Pictures, Inc., Hollywood, Calif.
"Development and Current Uses of the Acoustic Envelope"; H. Burris-Meyer, Stevens Institute of Technology, Hoboken, N. J. *(Demonstration.)*
"Notes on the Mechanism of Disk Recording and Playback"; O. Kornei, Brush Development Co., Cleveland, Ohio.
"Analytic Treatment of Tracking Error and Notes on Optimum Pick-up Design"; H. G. Baerwald, Brush Development Co., Cleveland, Ohio.

Adjournment.

REGISTRATION

Convention registration and information headquarters will be located on the Sagamore Hotel roof, adjacent to the *Glass House*, where all technical sessions and symposiums will be held.

Members and guests attending the Convention will be expected to register, and so help to defray the Convention expenses. Convention badges and identification cards will be provided for admittance to all regular and special sessions during the Convention. The identification card will also be honored through the courtesy of Loew's Theaters, Inc., at Loew's *Rochester* Theater and, through the courtesy of Monroe Amusements, Inc., at the *Palace*, *Regent*, and *Century* Theaters.

TECHNICAL SESSIONS

All the technical sessions of the Convention will be held in the *Glass House* on the roof of the Sagamore Hotel with the exception of Wednesday morning and evening, as described below. Members should note that the banquet, which at past conventions has always been held on Wednesday evening, this time has been scheduled for Tuesday evening to permit holding a special meeting on Wednesday evening at the *Eastman* Theater.

Wednesday, May 7th, will be devoted to a joint meeting of the Acoustical Society of America and the SMPE, consisting of a symposium of papers by engineers of the Bell Telephone Laboratories in the morning and afternoon. In the evening a demonstration of stereophonic sound will be given by the Bell Telephone Laboratories at the *Eastman* Theater.

LUNCHEON AND BANQUET

The usual Informal Get-Together Luncheon for members, their families, and guests will be held in the *Starlight Room* on the hotel roof on Monday, May 5th, at 12:30 P. M. Luncheon tickets should be procured when registering.

The 48th Semi-Annual Banquet and Dance will be held in the *Starlight Room* on the hotel roof on Tuesday evening, May 6th, at 7:30 P. M.: music and entertainment. Banquet tickets should be procured and tables reserved at registration headquarters by noon of Tuesday, May 6th.

LADIES' PROGRAM

Mrs. C. M. Tuttle, *Convention Hostess*, and members of her Committee are arranging a very attractive program of entertainment for the ladies attending the Convention. A reception parlor will be provided for the use of the Committee during the Convention.

Bell Telephone System
TECHNICAL PUBLICATIONS

ACOUSTICS

MONOGRAPH
B-850

A HIGH SPEED LEVEL RECORDER FOR ACOUSTIC MEASUREMENTS

BY

E. C. WENTE, E. H. BEDELL
and K. D. SWARTZEL, JR.
Bell Telephone Laboratories

DESCRIPTION OF A SYSTEM FOR CONTROLLING AND RECORDING THE GAIN OF AN AMPLIFIER IN ACCORDANCE WITH CHANGES IN THE INPUT POWER

Presented before
ACOUSTICAL SOCIETY OF AMERICA
WASHINGTON, D. C. MAY, 1933
Published in
THE JOURNAL OF THE
ACOUSTICAL SOCIETY OF AMERICA
VOL. VI, PP. 121-129 JANUARY, 1935

A High Speed Level Recorder for Acoustic Measurements

By E. C. WENTE, E. H. BEDELL and K. D. SWARTZEL, JR.,
Bell Telephone Laboratories

TWO quite accurate means for recording rapid variations in sound intensity in a form suitable for visual inspection have been available for a number of years. One of these is the phonodeik,[1] or one of its variants, and the other is a combination of a microphone and an oscillograph. When properly designed these devices record the actual wave form of the sound. However, for many acoustic measurements, a knowledge of the wave form is of secondary interest, whereas it is important that one should be able to record rapidly varying mean intensities over a wide range of values. From a record of the wave form it is not easy to determine the intensity with any degree of accuracy for a range greater than twenty or thirty db, but in some types of acoustic measurements it is highly desirable that the record cover a range of at least sixty db. Recently several types of instruments have been built which record, on a logarithmic scale, the mean power of the electrical input.[2,3,4] These instruments, like that described here, may be used to plot the intensity level in db as a continuous function of either time, frequency, or any other variable. The adaptability of such level recorders to acoustic measurements depends, among other factors, upon the range and accuracy of the logarithmic scale, and upon the effective speed of the recording mechanism. This recording speed is most conveniently expressed in terms of the rate, in db per second, at which the recorder is capable of following changes in the input power.

The level recorder described here consists essentially of an amplifier and rectifier, the output current of which is held at a substantially constant value automatically by a change in the gain of the amplifier, following changes in input power. The gain is varied by means of motor driven slide wire potentiometers graduated in logarithmic steps, the gain settings of which are recorded.

A schematic drawing of the recorder is shown in Fig. 1. Two potentiometers are provided in adjacent stages of the amplifier so that

[1] D. C. Miller, "The Science of Musical Sounds," p. 78.
[2] F. H. Best, *Bell Syst. Tech. J.*, Jan. 1933.
[3] L. Fenyo, *T. F. T.*, Jan. 1933.
[4] Ballantine, *J. Acous. Soc. Am.*, 5, 10 (1933).

by means of a selective switch either or both may be connected in the amplifier circuit. The potentiometers have sixty contact blocks, thus providing sixty distinct steps in the gain adjustment. The resistances between the various contacts are so graduated that each step produces the same number of decibels change in gain. In our measurements we have found two potentiometers particularly useful, one having a total range of sixty db, with steps of one db, and the other a range of thirty db, with steps of one-half db. A choice of either of the three ranges, thirty, sixty or ninety db, is thus immediately available for the ordinate scale. The greater ranges are more suitable for reverberation time or intensity range measurements, while the smaller, with

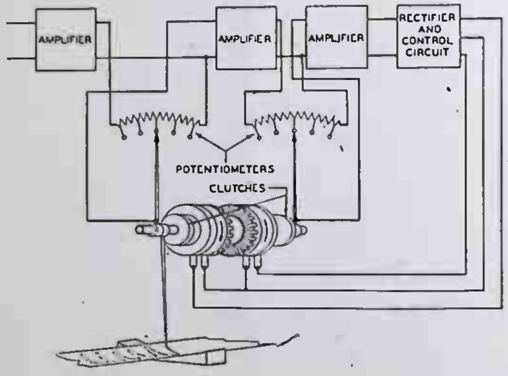

Fig. 1—Schematic drawing of level recorder.

correspondingly smaller steps are preferred where values of greater precision are desired. The potentiometers, however, are so constructed that they may be easily removed and replaced by others if other ranges are required.

The paper used in the recorder is an inexpensive standard commercial product. It consists of a colored paper with a white wax coating on one side, and the record is made by a stylus which scratches through the wax coating, thus producing a dark line on a white background. The records are permanent and may be photographed. Rectangular coordinates are obtained from the circular motion of the stylus by running the paper strip over a guide in the form of an arc, having a radius of curvature equal to the length of the stylus arm. This guide is placed slightly above a plane formed by the paper driving roller and a secondary guide, so that no constraint, other than the tension in the paper is required to hold it against the concave side of the guide. The width of the paper strip is $2\frac{1}{4}$ inches, of which 2 inches are used for

recording, so that the ordinate scale is 15, 30, or 45 db per inch, depending upon which potentiometers are connected in the amplifier circuit.

The potentiometer contacts and the recording stylus are moved back and forth by means of a magnetic clutch mechanism. Two mushroom shaped electromagnets are rotated about their common axes in opposite directions by a pair of bevel-gears driven from a single pinion. Disk shaped armatures ride on the polar ends of these magnets. These armatures are fastened to a shaft which passes through the centers of the magnets. To this shaft are also attached the potentiometer contact and stylus arms, and a pair of limiting switches which short circuit the appropriate clutch winding when the contactor and recording stylus reach either extreme position. When one of the magnets is energized it carries its armature with it, and so rotates the shaft in one direction, while, if the other magnet is energized it rotates the shaft in the opposite direction. The armatures are at all times pressed lightly into contact with the polar ends of the magnets in order to get the maximum speed of operation.

The potentiometer sliding contactors and the recording stylus are each carried on arms which are in the form of hollow truncated cones. This form was adopted in order to get the requisite strength with a low value of moment of inertia. The total moment of inertia of the moving system which carries these arms is less than 20 g $cm.^2$ The stylus has a rounded steel point which is pressed down on the paper by a small cantilever spring with a force just sufficient to penetrate the wax coating. The wax coating provides a uniform resistance to the motion of the stylus, and so reduces the possibility of free oscillation by preventing over-travel of the potentiometer sliding contacts.

The paper is moved, by means of rollers, through a train of gears, the speed reduction of which is adjustable. The possible paper speeds afforded by these gears varies from three inches per second to one-half inch per minute. The clutch pinion is driven through another similar train so that a choice of recording speeds is independently available. This recording speed, which for a level recorder such as described here, is simply the rate at which the amplifier gain can be changed to follow changes in the input power, is proportional to the total range of the potentiometers being used. With the train of gears ordinarily employed and the ninety db ordinate scale, four recording speeds, differing by factors of two, from 70 db per second to 560 db per second are available. By means of other gear ratios and potentiometer ranges the range of recording speeds has been extended to from 4 db per second to 850 db per second. The higher speeds are necessary in determining the power variations in speech or music, or in measuring

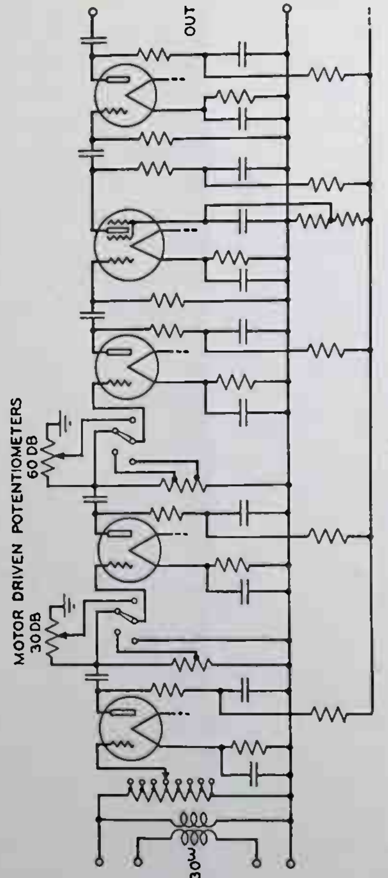

Fig. 2—Electrical circuit of level recorder amplifier.

very short reverberation times,[5] while the lower speeds are preferable when it is desired to average out the rapid variations in the input power.

The electrical circuit of the recorder amplifier is shown in Fig. 2. The input transformer has a low impedance winding designed for use with a moving coil microphone, but a high impedance terminal is also provided. All of the amplifying tubes, except the fourth, which is a high gain screen grid type, are of a three element heater type selected for a low noise level. The motor driven gain control potentiometers form part of the coupling between the first and second and the second and third stages. Manually operated gain controls are also provided, one at the input of the amplifier, and others which can be used in place of either of the motor driven potentiometers. The greater part of the

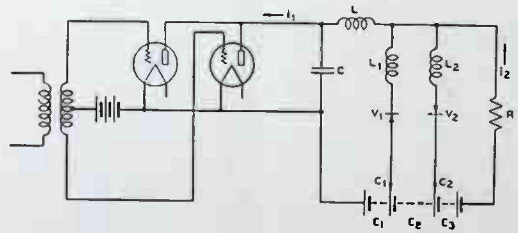

Fig. 3—Simplified rectifier and control circuit.

gain of the amplifier follows the automatic gain controls. This fact requires that the noise introduced by the operation of the potentiometers shall be very low, but it also eliminates the necessity of having a large power capacity in the tubes ahead of the potentiometers. The noise introduced by the operation of the potentiometers is reduced by the use of condensers having a very low direct-current leakage in the plate circuits of the tubes ahead of the potentiometers. The amplifier output is rectified by two pentode type tubes connected in a push-pull circuit. The recorder is operated entirely from a 60-cycle power line, the amplifier voltages being supplied from a power pack of conventional design. A small synchronous motor drives the gear trains for both the paper roller and the clutch pinion.

In an earlier form of recorder of this same general type the control circuit included a high speed mechanical relay, which was later replaced by a circuit containing gas filled grid control (or thyratron) tubes. The rectifier and control circuit shown in Fig. 3, which is now used, has been found to be highly satisfactory, and affords a convenient

[5] The application of this instrument to reverberation time measurements is discussed in another paper (*J. Acous. Soc. Am.*, Jan. 1935).

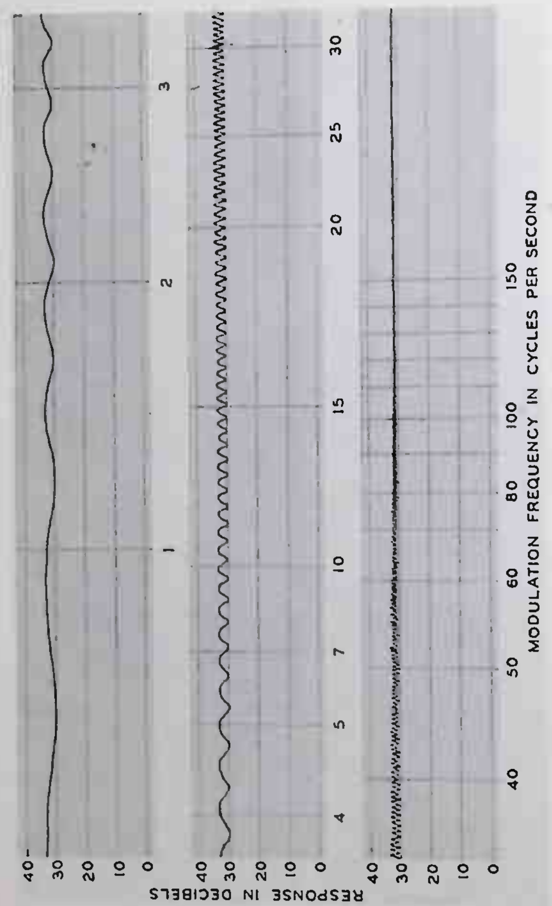

Fig. 4—Curve obtained for a modulated 1000-cycle input voltage with the modulation frequency continuously increased.

means of adjustment. L_1 and L_2 represent the magnetizing windings of the clutches. V_1 and V_2 are rectifiers of the copper oxide type, conductive in the directions indicated by the arrow heads. When the current i_2 is less than E_3/R current will flow through L_2; and when it is greater than $(E_2 + E_3)/R$ current will flow through L_1. The clutches are so connected in the circuit that when current flows through L_2 the potentiometer contacts are moved so as to increase the gain of the amplifier, whereas, when current flows through L_1 they are moved so as to reduce the gain. The range of the current i_2, for which neither of the clutches is energized sufficiently to move the contactors, can be controlled by adjustment of the voltage E_2. With the apparatus as

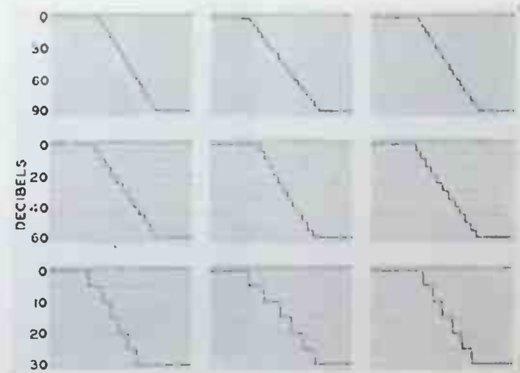

Fig. 5—Calibrations of the 30-, 60- and 90-db ranges in five db steps.

constructed this range can be reduced to zero, but if it is made much smaller than the change in gain of one step in the potentiometer setting the system will oscillate even though the input is constant. In practice this range is reduced to a point just short of the condition of instability. The inductance, L, and the condenser, C, are placed in the circuit as shown, to filter out partly the ripples in the rectified current, but the action of this filter must not be so effective as to level off variations in the power input that one may wish to record. The voltages for the control circuit are supplied from the same power pack as the amplifier voltages, rather than from batteries as shown in Fig. 3.

A recorder to be used for recording rapid variations in powers, such as those of speech or music, should not only be capable of following rapidly varying levels when such variations are in one direction, i.e., continually increasing or decreasing, but it should also be capable of following rapid changes in the direction of such variation. It was for this reason that the moment of inertia of the moving system of the recorder was kept at a low value. In Fig. 4 is shown a record obtained when a modulated current of 1000 cycles was applied to the recorder with the difference between the maximum and minimum values of the

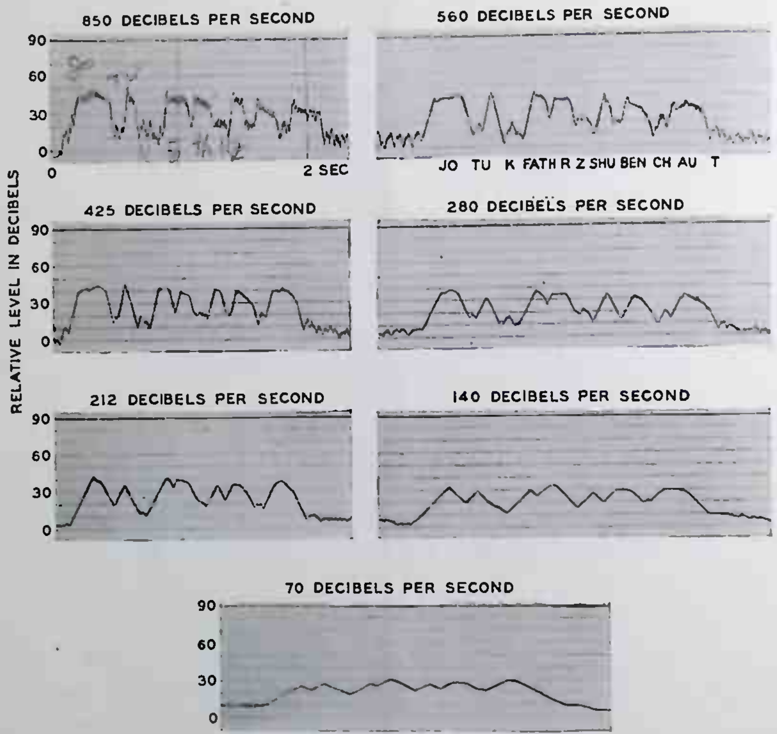

Fig. 6—Level variations in speech at different recording speeds.

current equal to 3 db. This record, and some others to be shown, was taken with a single motor driven potentiometer, having a range of 40 db, in the amplifier circuit. The modulating frequency was increased continuously until the recorder was no longer able to follow the variations. This record shows that the recorder is capable of following fluctuations in power up to a value which lies beyond one hundred and fifty per second. In some measurements it is, of course, more desirable to average the power over a longer period of time, the rapid fluctuations being of no particular interest. For obtaining such

data the recorder may be operated with the clutch pinion running at a lower speed.

Calibrations of the three ranges available for the ordinate scale when the 30 db and 60 db potentiometers are used are shown in Fig. 5. In each case the input to the level recorder was progressively changed in steps of 5 db until the full range had been covered. The fact that potentiometers of the type used are accurate over a wide range of frequencies is illustrated by the three sets of calibrations given in Fig. 5. Those on the left-hand side were taken for an input voltage having a frequency of 20 cycles, those in the center with 1000 cycles, and those on the right for 17,000 cycles.

In Fig. 6 some records of speech power are shown. The sentence, "Joe took father's shoe bench out," which is used in transmission tests, was spoken into the microphone. The seven records shown are for the same speaker who repeated the sentence at approximately the same rate and with the same inflection. The recording speeds, in db per second, are given on the curves. The records taken at the higher speeds show how the recorder may be used to measure syllabic power in speech, while those taken at the lower speeds give the power average over longer time intervals. Figure 7 shows some curves similar to those of Fig. 6 except that the source of sound was music from a piano. Both of the above groups of curves illustrate the action of the recorder as a recording volume indicator.

The upper two curves of Fig. 8 show the decay in the emission of sound from a tuning fork. In the first case the fork was undamped and the curve shows a rate of decay of 0.57 db per second. In the second case the fork was damped by a felt pad between the prongs and the curve shows a damping of 4.0 db per second. The lower curve in Fig. 8 is a record of street noise taken with the paper running at a relatively low speed.

The curves in Fig. 9 show how the recorder may be used in plotting response-frequency curves for loudspeakers or microphones. For the upper curve the microphone was placed about two feet from a loudspeaker in a room having a volume of about 10,000 cubic feet. The lower curve was obtained in the same room with the microphone about eight feet from the loudspeaker and shows the effect of the greater ratio of reverberant to direct sound in producing irregularities in the curve. An auxiliary stylus is provided for marking frequencies, or time, or other reference points on curves such as these, and those given in Fig. 10.

The first curve on the upper record of Fig. 10 is the combined frequency characteristic of the level recorder itself and of a heterodyne

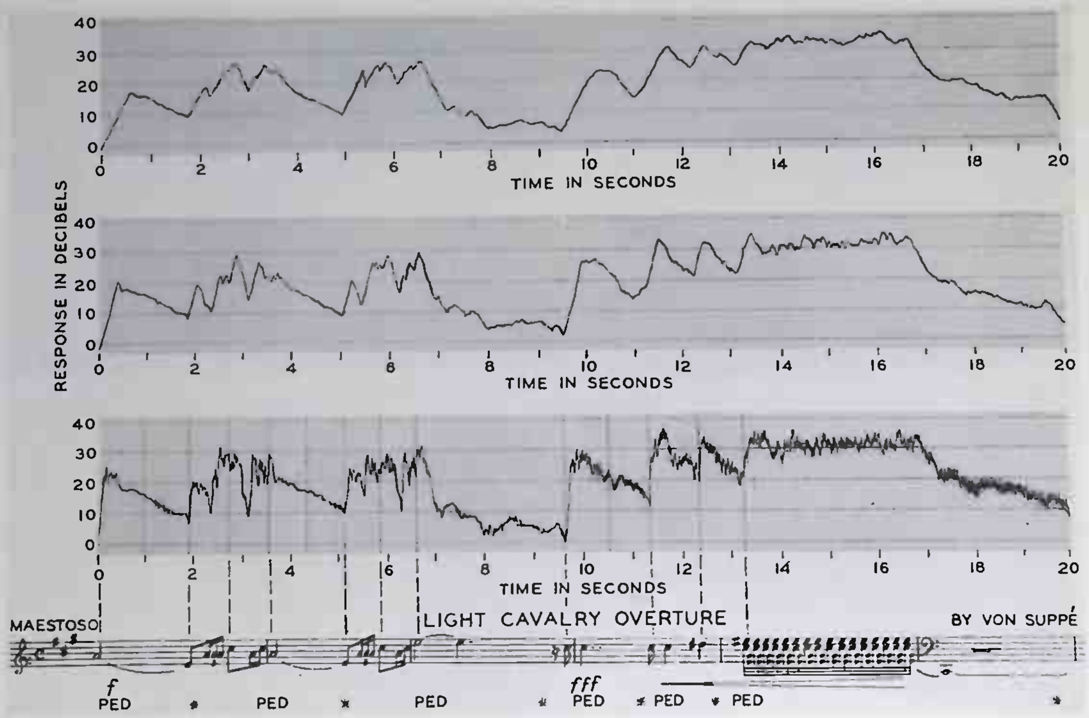

Fig. 7—Level variations in piano music.

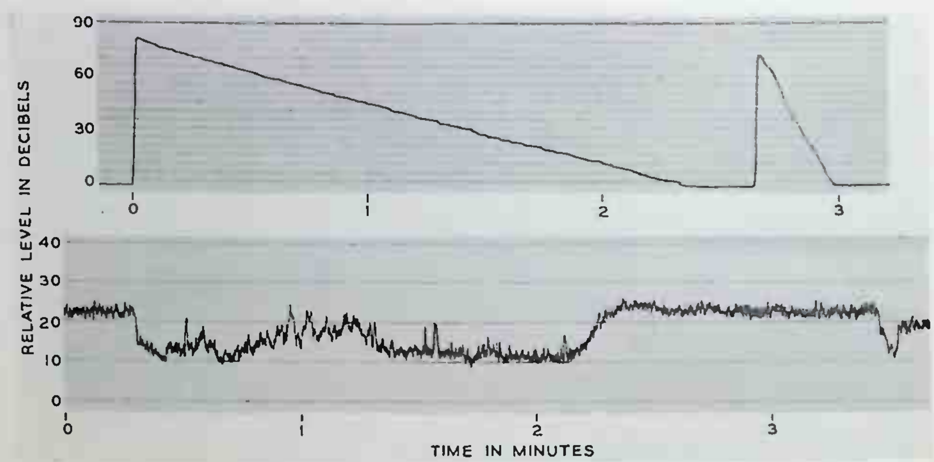

Fig. 8—Decay of an undamped and damped tuning fork, and a record of street noise.

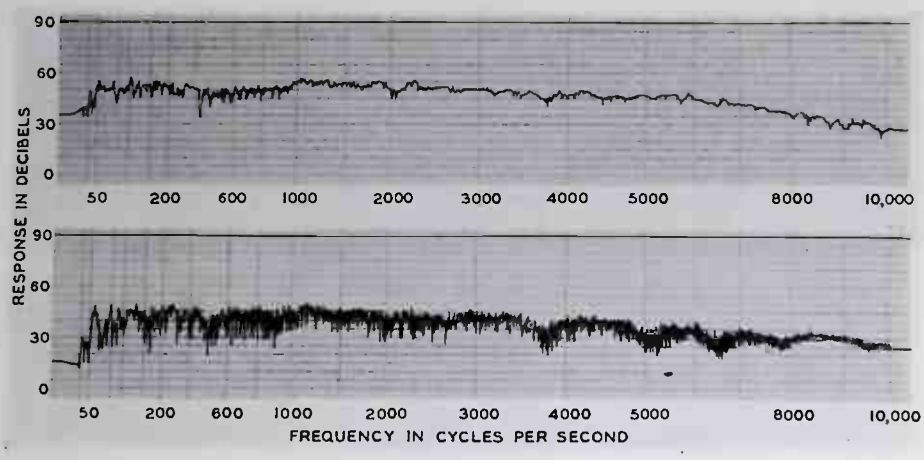

Fig. 9—Loudspeaker and microphone response curves taken at different distances in the same room.

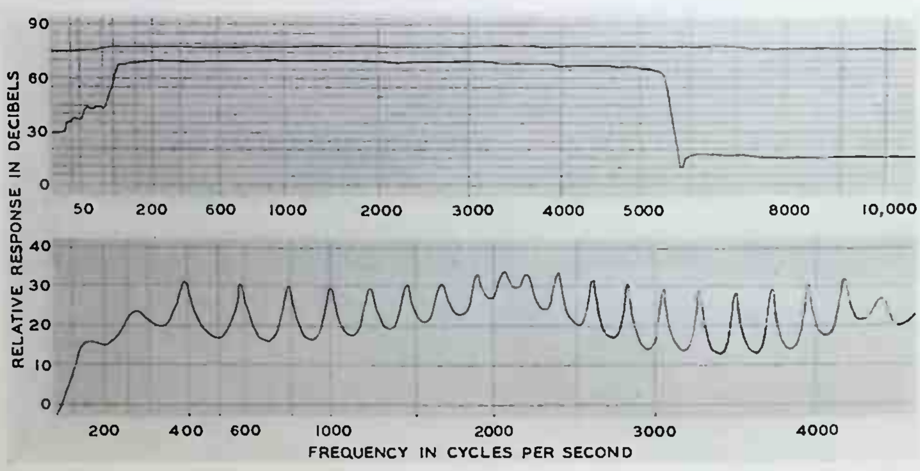

Fig. 10—Upper record: Output characteristic of heterodyne oscillator, and transmission characteristic of a 100–5500-cycle band-pass filter. Lower record: Harmonic resonance in an air column.

oscillator connected to it. This curve shows that the oscillator output and the level recorder amplifier are nearly enough uniform so that no corrections ordinarily need be made in response frequency characteristics taken with them. The second curve on this record is the transmission frequency characteristic of a transmission line which includes a 100-cycle to 5500-cycle band pass filter. The transmission characteristics of various types of electrical circuits can be easily obtained in this manner. The lower record in Fig. 10 is the transmission character-

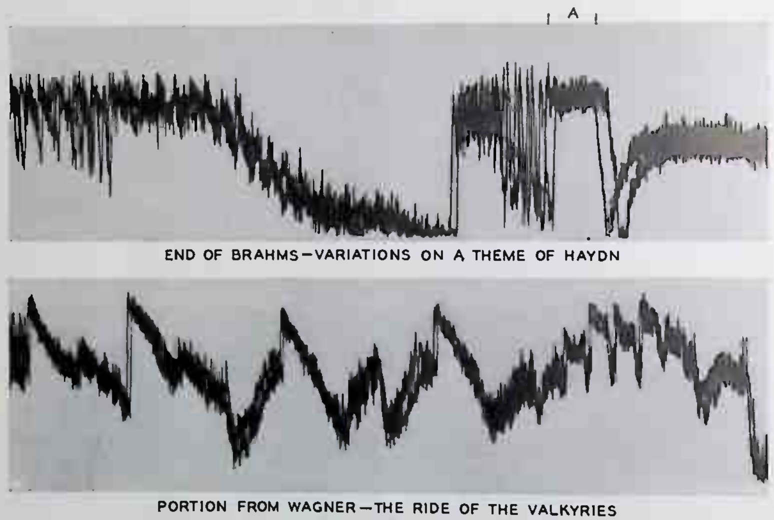

Fig. 11—Portions of superimposed records of orchestral music played on successive days.

istic of an acoustic line consisting of a three inch tube, about three feet long, having a driving unit and a microphone loosely coupled to the two ends. The harmonic resonance in the tube is clearly shown.

The two curves shown in Fig. 11 are portions of a record of the intensity level of the music from the Philadelphia Symphony Orchestra, playing in the Academy of Music in Philadelphia, under the direction of Leopold Stokowski. This record was first taken during a matinee concert. The paper was then rerolled and a second record was taken, superimposed on the first, during an evening concert, with the same musical program, given on the following day. The total range of the ordinate scale on these curves is 40 db, the recording speed was 240 db per second, and the time scale is approximately one foot per minute.

These curves show that intensity changes of the order of 40 db may occur during relatively short passages of music, and for some selections, intensity ranges of 75 db were measured. The portions of the records shown in Fig. 11 are interesting from another standpoint, however. They show that the tempo of the music is maintained quite accurately throughout a long concert, and also that the intensities are reproduced

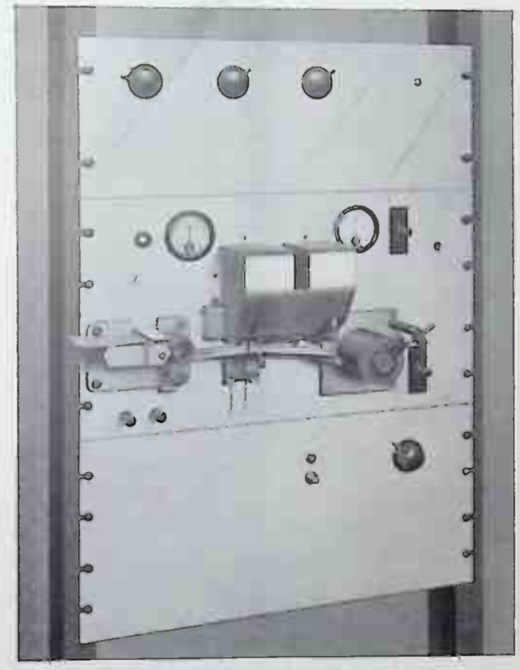

Fig. 12—Photograph of the complete recorder as mounted on a relay rack.

within very narrow limits. This is particularly noticeable in the lower curves of Fig. 11 where the intensities of the two records seldom differ by more than one or two db, although there are several abrupt changes of the order of 20 db. In the upper curves of Fig. 11 the music ends just after the interval marked "A." The intensity level of the applause from the audience, which begins almost immediately after the end of the music is shown by the last portions of the upper curves.

The above records are illustrative of the many types of acoustic and electrical measurements for which the recorder is applicable. Figure oscillator connected to it. This curve shows that the oscillator output and the level recorder amplifier are nearly enough uniform so that no corrections ordinarily need be made in response frequency characteristics taken with them. The second curve on this record is the transmission frequency characteristic of a transmission line which includes a 100-cycle to 5500-cycle band pass filter. The transmission characteristics of various types of electrical circuits can be easily obtained in this manner. The lower record in Fig. 10 is the transmission character-

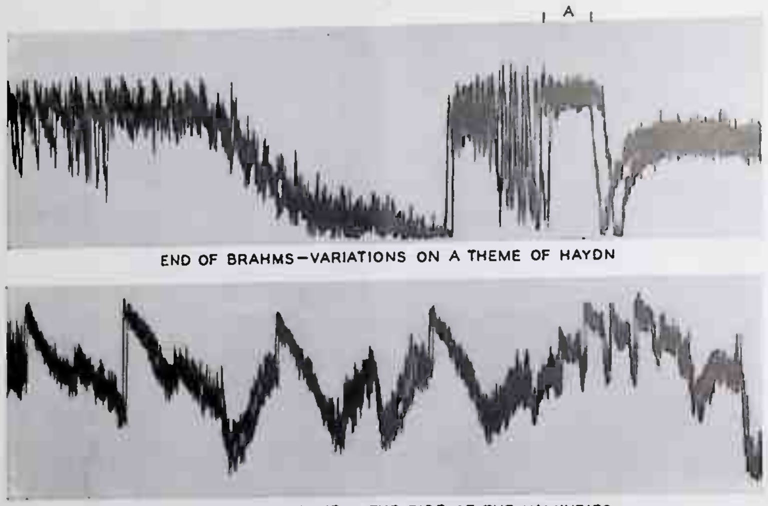

Fig. 11—Portions of superimposed records of orchestral music played on successive days.

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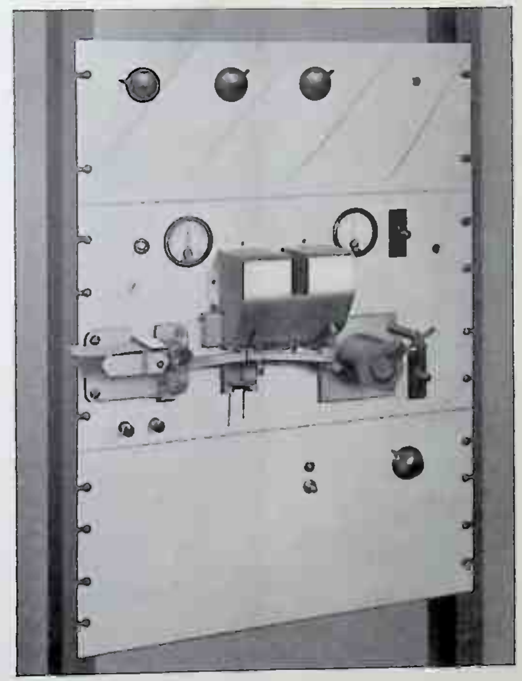

Fig. 12—Photograph of the complete recorder as mounted on a relay rack.

within very narrow limits. This is particularly noticeable in the lower curves of Fig. 11 where the intensities of the two records seldom differ by more than one or two db, although there are several abrupt changes of the order of 20 db. In the upper curves of Fig. 11 the music ends just after the interval marked "A." The intensity level of the applause from the audience, which begins almost immediately after the end of the music is shown by the last portions of the upper curves.

The above records are illustrative of the many types of acoustic and electrical measurements for which the recorder is applicable. Figure

12 is a photograph of the complete recorder as it is mounted on a rack. The amplifier, the mechanical system, and the rectifier for the power supply, each occupy one of the three panels, and the panels are interconnected with plug-in cords so that the instrument is readily portable.

BELL TELEPHONE LABORATORIES
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THE REPRODUCTION OF ORCHESTRAL MUSIC IN AUDITORY PERSPECTIVE

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THE REPRODUCTION OF ORCHESTRAL MUSIC IN AUDITORY PERSPECTIVE

As Demonstrated before the
National Academy of Sciences,
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Washington, D. C.

IN THIS electrical era it does not seem very remarkable that music, even from a large symphony orchestra, should be picked up by microphones, transmitted over telephone wires, and reproduced at a distant point. Most of us probably hear it accomplished every day by means of the radio, and radio transmission and reproduction would, in general, be called good. Between good reproduction and perfect, however, there is a very wide gap, and the difficulties of crossing it are probably not realized by those not technically familiar with the subject. Perfect reproduction, of a symphony concert for example, would make it impossible for one listening with his eyes blindfolded to know that the actual orchestra was not on a stage before him. Not only would every tone and overtone be present in its correct relative volume, but there would be a depth and color not ordinarily obtained when electrical apparatus intervenes between orchestra and audience.

Three classes of requirements must be met if the reproduced sound is to be indistinguishable from the original. Two of them, that both the complete frequency and complete volume ranges be transmitted, have been generally recognized for some time. The third, that the sounds must be reproduced with the correct auditory perspective, has been fully appreciated only by those most closely associated with the science of sound reproduction.

Sounds in general are composed of a group of tones and over-tones ranging from the deep bass of the lowest organ notes, or those of a bass drum, to the shrillest tones the ear can hear. Each note of a musical instrument has a fundamental tone and a group of harmonics. The fundamental tone sets the pitch, and the harmonics give the note its quality. It is the harmonics that make it possible to distinguish a note on a violin from one on a trumpet or from any other of the same pitch. It is in the harmonics that reside the richness of music and the wealth of sensuous appeal. These tones and over-tones are known and recognized by their frequency, or vibratory rate; and the range of frequencies to which the ear responds runs from about 16 cycles per second to 16,000, or even 20,000 cycles for some ears. The sensitivity of the ear falls off rapidly at the higher frequencies, however, so that the effect of frequencies above 15,000 cycles is negligible for the most part. The highest note on the piano has a fundamental frequency of only about 4,000 cycles, and few of the musical instruments exceed this pitch, but the accompanying harmonics, or over-tones, which are of still higher frequencies, are very necessary to the proper quality and richness of the notes.

Of no less importance, if the full aesthetic effect of music is to be obtained, is the range in volume. The ear has a recognizable range of volume as it has of frequency.

This extends from sounds so low that the ear cannot hear them, to sounds so great that the sensation is one of pain rather than of hearing. For convenience in scientific study, the power of sounds is graded in units known as decibels (abbreviated db). The threshold of hearing is taken as a reference base, and the ordinary audible range runs from the volume of sound one would hear in a quiet garden, or that of an average whisper at a distance of four feet, which are at a level of 20 db, to that of a pneumatic riveter, at a level of 100 db—a total range of about 80 db. The range of a large symphony orchestra is about 70 db, so if the music of such an orchestra is to be faithfully transmitted electrically, a volume range of the order of 70 db must be transmitted: a range of power of ten million to one.

The third requirement becomes of particular importance when the sound to be transmitted and reproduced is that from a large and relatively widely spaced group of instruments, such as a complete symphony orchestra. When one sits in an auditorium and listens to a symphony concert he experiences something that is over and above the effect produced by the actual frequency and volume range given out by the orchestra. This additional appeal is difficult to describe, and almost impossible to measure. It is partly due to a spreading of the sound in all directions so that it fills the entire volume of the auditorium and thus reaches one's ears by various paths. It is partly due to other factors; but whatever its cause it results in a richness and texture of tone that no ordinary electrical reproduction can provide. For lack of a better term, the effect may be called auditory perspective. Without it the music would be one dimensional and not expanded into its true spatial relationship. The difference may be compared to that between the appearance of a photograph of a scene and the same scene when viewed through a stereoscope.

How to obtain this auditory perspective in music transmitted and reproduced electrically was discovered by the scientists of Bell Telephone Laboratories as a result of their fundamental investigations in acoustics and telephonic transmission. During the course of those investigations they had developed telephonic systems of high quality, but for their further researches they needed opportunity to utilize music in its most perfect forms. Now it happened that Dr. Leopold Stokowski, Director of the Philadelphia Orchestra, was interested in the possibilities of electrical systems for the production of exceptional orchestral effects. Through his voluntary cooperation, therefore, the Laboratories' scientists were able to make quantitative physical studies of music as rendered by his orchestra, and so to perfect their designs; and with the completion of the new equipment some of the possibilities which Dr. Stokowski had hoped for became practicable. An extended series of tests was then carried on in Philadelphia in which the Laboratories' scientists were generously assisted by Dr. Stokowski. As a result of these studies, it was found that by employing two microphones, one properly located on each side of the stage, and by transmitting over two separate circuits to two of the newly developed loud speakers, similarly placed, the effect of the actual presence of the orchestra could be obtained.

Even with the discovery of a comparatively simple

means of obtaining true auditory perspective, the problem was not completely solved. Never before had either the complete frequency, or the complete volume range, of a symphony orchestra been commercially transmitted and reproduced. No complete chain of apparatus, from microphone to loud speaker was available, that would faithfully transmit the entire range of frequency and volume. Microphones perhaps offered the fewest difficulties. Bell Laboratories had already designed sensitive microphones that would transmit practically the entire range required, and only minor modifications were needed to make them entirely suitable.

This was not true of the amplifiers. There had to be developed amplifiers which would faithfully transmit all frequencies from 35 to 16,000 cycles at levels from the barely audible pianissimo effects to the resounding orchestral crashes of ten million times greater power; and all the pieces of apparatus had to be so designed that even during intervals of complete silence not the slightest noise would be introduced to suggest the presence of electrical apparatus. No underlying hum or noise such as is commonly present in radio or other systems of reproduction, could be tolerated with the new apparatus. In the intervals of silence there must be real silence: a dead auditory void in which the fall of the lightest pin could be heard. This has actually been accomplished to a degree heretofore unknown. Probably the most quiet electrical reproduction up to the present is that obtained with high-grade sound picture apparatus; but such apparatus at its most quiet moments gives off some 300 times more sound power than the new apparatus when all the musicians are silent.

Of even greater difficulty possibly was the design of suitable loud speakers. It is not practicable to obtain the entire frequency range with a single unit, and so two types of loud speakers are used. One, somewhat resembling the horns used for sound pictures, is employed for the frequencies from 35 to 300 cycles; and another type, for the range from 300 to 16,000 cycles. These loud speakers are different from anything previously produced commercially. Never before have these elements fulfilled such difficult requirements of frequency range and volume. The best sound picture systems record and reproduce approximately half the range of frequencies handled by the new loud speakers, and the best radio systems even less. In volume range the comparison is equally remarkable. Although sound picture systems under the most favorable conditions may provide a volume range of 40 or 45 db, radio systems rarely exceed 30, while the range provided by the new apparatus is well above 80. Whereas the power range of radio is of the order of 1000 to 1, the new equipment is capable of yielding a range of 100,000,000 to 1.

The new loud speakers and their associated equipment of amplifiers and microphones are, therefore, fully capable of handling the entire volume range of a symphony orchestra. When one speaks of range of loudness which can be handled by an electrical system for reproduction, one is concerned with the differences between the loudest and faintest passages of the music which it can reproduce. There is in addition the problem of handling the peaks of maximum loudness. These peaks in the case of music from

a symphony orchestra are beyond the possibilities of the ordinary loud speaker to reproduce without distortions which seriously affect the musical sonority. The low frequency sounds make the largest contribution to the peaks of sound power which must be handled to meet these conditions. The diaphragm of the low frequency element in the new loud speaker has been made nearly seven times larger than that of the elements used ordinarily for sound picture reproduction. By these diaphragms a large column of air is set into motion.

The ordinary loud speaker also becomes directional in its characteristics at the higher frequencies. Low frequency sounds spread in all directions from the mouth of the horn, but the higher frequencies tend to concentrate into a beam projected directly ahead of the horn; and the width of the beam becomes narrower and narrower as the frequency increases. Because of this fact, the audience in a large hall equipped with the ordinary loud speakers, never hear quite the proper blending of frequencies. Those directly in front of the horn receive too great a proportion of the higher frequencies, while those on the sides receive too much of the low frequencies. To avoid this effect, the horn of each high-frequency element is divided into 16 diverging rectangular sections which spread the sound over an arc of 60 degrees vertically and one of 60 degrees horizontally. Two of these units placed side by side thus spread the sound over a horizontal angle of 120 degrees—a far wider coverage than has been obtained before and one which distributes the sound throughout the auditorium with a faithful blending of the frequencies.

Besides providing for the full volume range of the orchestra, the amplifiers have an additional amplification of at least 10 db, so that, if desired, the volume of loud passages may be made ten times as great as the actual output of the orchestra. Technically described, the maximum sound power of a symphony orchestra integrated over an interval of two-tenths of a second is less than 20 watts, whereas that possible from the loud speakers of the new apparatus is more than 200 watts. This additional gain allows effects to be obtained which have been impossible before. Besides the effects of range and quality of tone, the total aesthetic appeal of an orchestra is due in no small degree to the range in volume. The number of musicians one can place on a stage is limited. To put ten times as many as contained in a modern symphony orchestra is impossible in any existing hall. The control of volume given by the new apparatus enables the director to secure at will the equivalent of an orchestra of nearly a thousand musicians.

The advantage of this control of volume does not end here, however. Its presence makes it possible to reproduce operatic music, where a soloist is accompanied by an orchestra, without allowing the voice of the singer to be drowned out by the louder passages. For this purpose a third channel, including its separate microphone, transmission line and loud speaker, has been provided in the new system primarily for the singer. The volume of output of this channel is controllable independently of the other two. In this way the loudness of the voice may always be kept just above that of the orchestra and the desired

musical effect be obtained. There thus reside in the new apparatus possibilities heretofore unattainable; and telephonic research has laid a foundation for what may be one of the greatest advances in musical aesthetics of the present scientific era.

The first public demonstration of the new apparatus was given in Washington on the evening of April 27th under the auspices of the National Academy of Sciences. At that time Dr. Stokowski, Director of the Philadelphia Orchestra, manipulated the controls from a box in the rear of Constitution Hall, while the Philadelphia Orchestra, led by Associate Conductor Alexander Smallens, played in the Academy of Music at Philadelphia. Between Philadelphia and Washington, the music was transmitted over telephone cable circuits. The program consisted of the Toccata and Fugue in D Minor, of Bach; Beethoven's Symphony No. 5 in C Minor; L'après-midi d'un Faune, of Debussy; and the Finale of Götterdämmerung. A visual accompaniment was provided for the music by Electrical Research Products, Incorporated. Its stage direction—through the courtesy of the Yale School of Drama—was by S. R. McCandless, and the designs were by Eugene Savage and George Davidson.

During the intermission Dr. W. W. Campbell, President of the National Academy of Sciences, introduced Dr. Harvey Fletcher, Director of Acoustical Research at Bell Telephone Laboratories. With the assistance of the orchestra in Philadelphia, Dr. Fletcher then performed several experiments to demonstrate the important characteristics of the new apparatus. On the stage of the Academy of Music in Philadelphia, where the pickup microphones were installed, a workman busily constructing a box with hammer and saw was receiving suggestions and comments from a fellow workman in the right wing. All the speech and accompanying sounds were transmitted over cable circuits to the loud speakers on the stage of Constitution Hall in Washington. So realistic was the effect that to the audience the act seemed to be taking place on the stage before them. Not only were the sounds of sawing, hammering and talking faithfully reproduced, but the correct auditory perspective enabled the listeners to place each sound in its proper position, and to follow the movements of the actors by their footsteps and voices.

For another demonstration, the audience heard a soprano sing "Coming Through the Rye" as she walked back and forth through an imaginary rye field on the stage in Philadelphia. Here again her voice was reproduced in Washington with such exact auditory perspective that the singer appeared to be strolling on the stage of Constitution Hall.

An experiment which demonstrated both the complete fidelity of reproduction and the effect of auditory perspective was performed by two trumpet players. One, in Philadelphia at the left of the stage of the Academy of Music, and the other in Washington at the right of the stage of Constitution Hall but invisible to the audience, alternately played a few phrases of the same selection. To those in the audience there seemed to be a trumpet player at each side of the stage before them. It was not until after the stage was lighted that they realized that only one of

the trumpet players was there in person. The music of the other was transmitted from Philadelphia with such perfect fidelity and reproduced in such true perspective that it was impossible to tell that one of the players was absent.

The auditory perspective effect is not restricted to placing sounds in their correct positions across the stage, but is three dimensional. This was shown by having several sources of sound moved around the stage in Philadelphia, not only back and forth but high up in the center of the stage as well. The movement of each sound was faithfully reproduced by the loud speakers in Washington even when the sounds were carried high above the level of the stage floor.

To show the volume range possible with the new equipment, the orchestra played a selection at a constant level of loudness while the output of the loud speakers was varied from a level so low that the instruments could scarcely be heard, up to a loudness almost great enough to be painful. Throughout the whole range, the reproduction was faithful in all respects except the level of loudness; there was no distortion or noise to mar the perfection of the reproduction, and the wide range in volume was vividly impressed on the audience.

The effect of limiting the range in pitch, or frequency, was illustrated by employing electric filters to cut out one octave at a time—first from the upper end of the range and then from the lower. The new apparatus reproduces faithfully about 9 octaves or from 35 to 16,000 cycles, compared to about six for ordinary radio reproduction. By this demonstration the audience had the opportunity of judging the importance of the complete range to the full aesthetic appeal of music, and of comparing it with the more limited ranges ordinarily heard.

The technical features of these new developments, which were carried on as part of the research program of the American Telephone and Telegraph Company, were disclosed for the first time by Dr. F. B. Jewett, Vice President of the American Telephone and Telegraph Company in charge of development and research, at a meeting of the National Academy of Sciences on Tuesday afternoon of April 25th. In discussing the future of the new system, Dr. Jewett said in part:

"As to the future of the accomplishment shown here today, it is difficult to make any definite prediction. What we have done is to produce pickup microphones, amplifiers, electrical filters, transmission lines and loud-speaking reproducers so perfect that the entire frequency and volume range of the most exacting orchestral and vocal music can be reproduced at a distance without impairment of quality. We have also worked out the arrangements by which substantially perfect auditory perspective is possible. This latter is an essential part of the problem if realistic illusion as to the physical arrangement of the component parts of an orchestra is desired.

"We can place at the disposal of the musical director instrumentalities which will enable him to produce at a distant point, or at many distant points simultaneously, a completely faithful replica of the tonal effects produced locally in the auditorium on the stage of which the orchestra is performing. Likewise, portions of this same equipment place at his disposal the means of very greatly extending the range of orchestral reproduction and of making possible artistic effects hitherto unattainable.

"With these instrumentalities available, the questions of the manner and extent of their use are primarily questions for the musician and those interested in music rather than for the physicist and the engineer. Our job has been to produce a set of tools. The musicians and musical directors, and back of them the musical composers, must determine just how these tools can best be used and what they can best produce. By its very nature the ensemble of what we have created is primarily of value for musical production or reproduction in halls, theatres or auditoriums. In a word, its field of applicability is where a large number of people might congregate for the common enjoyment of music of distinction. In its present form it is not directly applicable to the limited environment of the home.

"These new tools offer not only an enlarged field of possibility to the musician and the composer for the production of auditory effects, but likewise a great broadening of the audience which derives pleasure from such effects. Many people, especially in our smaller cities, are now deprived of the ability to hear good orchestral music by the factors of cost and distance, and the element of time in going to the cities where orchestral music is normally produced. What we as physicists and engineers have done is to provide a mechanism for obviating these factors. Whether the results justify our expectations is for others to say."

BELL TELEPHONE

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ALTERNATING - CURRENT POTENTIOMETER

BY

E. C. WENTE

A GENERAL DESCRIPTION OF A
VACUUM-TUBE ALTERNATING-CURRENT POTENTIOMETER
FOR USE AT FREQUENCIES UP TO 10000 CYCLES PER SECOND

REPRINT OF PAPER

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A VACUUM-TUBE ALTERNATING-CURRENT POTENTIOMETER

BY E. C. WENTE

Research Laboratories of the American Telephone and Telegraph Company, and the Western Electric Company, Incorporated.

ABSTRACT OF PAPER

A description is given of an alternating-current potentiometer which is suitable for the measurement of electromotive forces having frequencies up to 10,000 cycles per second. Both the phase and magnitude are read directly from the dial settings.

A special type of a-c. differential galvanometer used in connection with the potentiometer is also described.

IN the investigation of many telephonic and acoustic problems it is necessary to measure very small voltages throughout a wide range of frequencies. Because of the outstanding advantages of the potentiometer method for measuring small electromotive forces it has seemed desirable to improve the means of applying it to this service. Numerous difficulties are encountered, however, in attempting to make use of the common types of alternating-current potentiometer circuits. The electromotive forces set up by stray magnetic fields from air-core inductance coils and the stray currents, which pass through the distributed capacities of the coils or through the capacities to ground of the different parts of the circuit, are sufficient to preclude the use of certain forms of apparatus. The Drysdale potentiometer, although relatively free from these sources of inaccuracies, does not cover the desired frequency range and has the further disadvantage of requiring a considerably greater a-c. input than the usual vacuum tube oscillator will deliver without a special amplifier. The form of a-c. potentiometer, which is here described, has been used successfully throughout the frequency range from 60 to 14,000 cycles per second, and the a-c. power required is so small that a standard type of vacuum-tube oscillator may be used as a source of alternating current.

A further advantage of the potentiometer over most of the other types that have been described is the fact that the readings give the phase and voltage of the measured e. m. f. directly. Since most a-c. potentiometer measurements are used in finding the ratio of electromotive forces, it is more convenient to have the readings given in this form than as the sum of two right angle component voltages.

The success of this type of potentiometer depends on the use of vacuum tubes not merely as amplifiers but as unilateral elements to prevent the undesirable reaction of some parts of the circuit upon others. It is thus possible to control the amplitude and phase of comparatively large currents without any troublesome

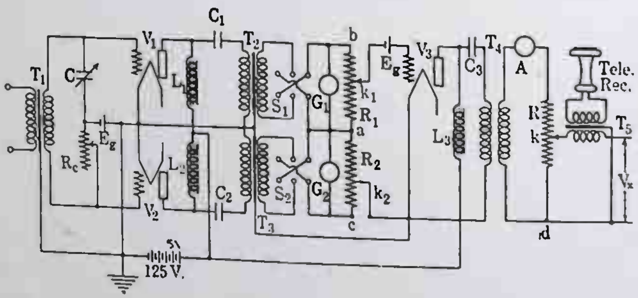

Fig. 1—Complete Potentiometer Circuit

reaction of the load on the controlling circuit. Special precautions have been taken to reduce the effects of stray magnetic fields and of mutual capacities to a minimum. For the sake of convenience in using this apparatus it has been found desirable to construct a new form of a-c. differential galvanometer and a special phase-shifting rheostat. With the potentiometer as constructed measurements of both voltage and phase may be made by a few simple adjustments, and experience has shown that the results are accurate over a wide range of frequencies.

The Circuit

The complete circuit of the potentiometer is shown in Fig. 1. V_1, V_2 and V_3 are three-element thermionic vacuum tubes of the audion type. The secondary of a transformer is connected to a condenser C, and a

resistance, R_c, in series. The input terminals of the vacuum tube, V_1 are connected across the condenser, and those of V_2 across the resistance. The output circuits of these tubes operate into the two transformers T_2 and T_3, the secondary windings of which are connected to two equal resistances, R_1 and R_2, respectively. By means of the sliding contacts, k_1 and k_2, part of the resistance R_1, and part of the resistance, R_2 are shunted across the input of the third vacuum tube. The output of this tube operates into a transformer, the secondary of which is connected to the potentiometer slide wire, R. T is either a telephone receiver or a vibration galvanometer, which serves the purpose of indicating when the drop of potential between d and k is the same in magnitude and phase as the unknown electromotive force V_x.

Theory of Operation

If the impedances, R_c and $\frac{1}{\omega C}$ are much smaller than the input impedances of the vacuum tubes, the differences of potential across the resistance, R_c, and across the condenser, C, will be practically 90 deg. out of phase. If R_1 and R_2 are equal, the transformers, T_2 and T_3, of identical design, and if the tubes V_1 and V_2, have the same output impedances, the currents in R_1 and R_2 also will be 90 deg. out of phase. By proper adjustment of C and R_c these currents may be made equal to each other in magnitude. To facilitate this adjustment, one coil of a differential a-c. galvanometer of special design is connected in shunt with R_1, and the other coil in shunt with R_2. By means of the sliding contact, k_1, any fractional part of the voltage drop along R_1, and by means of the sliding contact, k_2, any fractional part of the voltage drop along R_2, may be impressed on the tube, V_3. The two sliding contacts should be rigidly attached to the same dial so that their positions are varied simultaneously. These contacts are placed so that in one extreme position, the input terminals of the tube are connected across R_1 and in the other extreme position across R_2. The phase of the current in R may be varied through 90 deg. by changing the positions of these contacts between the extreme points. By the additional use of the reversing switches the phase may be shifted through 360 deg.

The resistances R_1 and R_2, must be designed so that the voltage impressed on the last tube is the same for all positions of the contacts. If $\frac{x}{l}$ is the fractional part of the distance k_1 is moved from a to b, and the fractional part of the distance k_2 is moved from c to a, then the resistance between a and k_1 should be

$$R \cos \frac{x}{l} \frac{\pi}{2}$$

and that between k_2 and a,

$$R \sin \frac{x}{l} \frac{\pi}{2}$$

The voltage between k_1 and k_2 will then be

$$I R \cos \frac{x}{l} \frac{\pi}{2} \sin p t + I R \sin \frac{x}{l} \frac{\pi}{2} \cos p t$$

$$= I R \sin \left(p t + \frac{x}{l} \frac{\pi}{2} \right)$$

where $I \sin p t$ and $I \cos p t$ are the currents in R_1 and R_2, respectively. Hence, in this case the magnitude of the voltage between k_1 and k_2 is constant and the scale of phase is linear. This arrangement then provides a convenient way of varying the phase of the output current of the last tube without varying its magnitude.

In making a measurement of the voltage, V_x, the phase of the current in R and the position of the contact, k, are adjusted so that no tone is emitted by the telephone receiver, T. If then I is the current flowing through the resistance, R, the magnitude of this voltage

is equal to $r I$, where r is the resistance between the points d and k.

The Phase-Shifting Rheostat

With the slide wires, R_1 and R_2, arranged as shown in Fig. 1, it would be necessary to have 900 points of contact if adjustments for phase were to be made to 1/10 of a degree, as is often necessary for obtaining

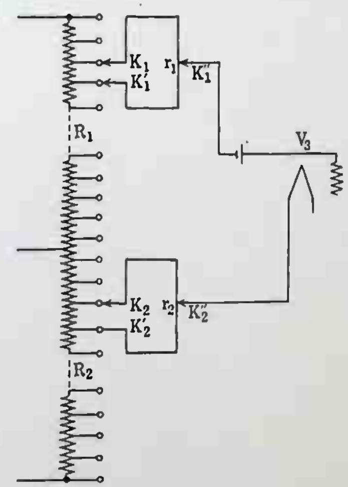

Fig. 2—Double Set of Contacts

complete silence in the telephone receiver, T_1. By the use of a double set of contacts as shown in Fig. 2, the construction is simplified. The sliding connectors, K_1 and K_1', make contact with two adjacent posts. These sliding connectors lead to the terminals of a secondary slide wire resistance, r_1, which should be large compared with that part of R_1 coming between any two adjacent contact posts. The sliding contacts, K_1, K_2, K_1' and K_2' are connected mechanically so that all four are moved simultaneously, likewise K_1'' and K_2''. The dial controlling the former set of contacts then serves for coarse adjustment of phase and that regulating the position of the latter for fine adjustment. In practise it has been found convenient to have 46 contact points on each R_1 and R_2, so that one step on the first dial shifts the phase through two degrees. With the second dial the phase adjustment may then be made to within a fraction of a degree.

The A-C. Differential Galvanometer

In Fig. 1 is shown how the two coils of a differential galvanometer are connected in the circuit to indicate when the potential differences across R_1 and R_2 are equal in magnitude. Most of the different types of instruments described in the literature which might be used for this purpose depend for their action on a heating element. However, hot wire instruments have

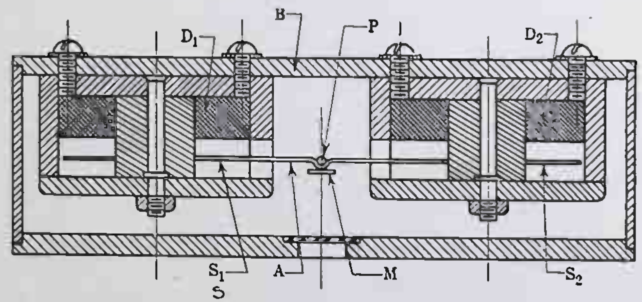

Fig. 3—Sectional View of Differential A-C. Galvanometer

to be readjusted frequently, burn out with a very small overload and have a comparatively low sensitivity. In general, they also have a low resistance and so are adapted to comparing currents rather than voltages. An electrodynamic type of differential galvanometer was therefore designed, which gave entirely satisfactory results.

Fig. 3 will make clear the construction of the instrument. To the frame, B, are attached two small shell-type transformers, of which D_1 and D_2 are the primary windings. The secondary windings, consisting of the aluminum rings, S_1 and S_2, *i. e.*, of single short-circuited turns, are rigidly held together by the bar, A, and thus form with the mirror, M, the moving system suspended at P.

When an alternating potential difference is applied to one of the primary coils a force of repulsion will be

set up between this coil and the corresponding secondary winding. If then the two transformers are exactly alike and the point of support, P, is midway between the coils, S_1 and S_2, there will be no rotation of the moving system from its normal position whenever the electromotive forces impressed on the primary windings are of the same frequency and are equal in magnitude. The instrument thus affords a convenient method for determining the equality of two voltages of the same frequency.

Neither the impedance nor the sensitivity of this instrument is constant with frequency. But this defect is of no great consequence so long as the use of the instrument is confined to a comparison of voltages of the same frequency. This case, in fact, is the only one where this particular type of differential galvanometer has any practical value. Here, however, it has the following points to commend it. The primary coils may be wound to any desired impedance so that the galvanometer can be adapted for use in circuits carrying either large or small currents. The sensitivity appears to be several times greater than that of the various hot-wire types of instruments. The instrument may be subjected to a very large overload without injury. The moving system has exceptionally good mechanical characteristics, short period and high damping. An advantage of this type of instrument over one that might be constructed on the principle of the ordinary electrodynamometer, is the fact that no lead wires are required for sending current through the coils of a moving system. The mechanical construction is therefore much simpler in this respect.

Sources of Inaccuracies

Since one of the output terminals of a vacuum-tube oscillator is generally either completely grounded, or has a comparatively low impedance to ground, the mutual capacity between the primary and secondary windings of the transformers, T_1, should be small, otherwise the condenser, C, and the resistance, R_c, will not be traversed by the same current. In this case, the potential difference across the condenser and the resistance will not necessarily be 90 deg. out of phase. Inaccuracies on this account may be kept small if the transformer has a grounded shield between its windings and if the capacity between this shield and the secondary winding is small compared with the capacity of the condenser, C. In order to have the phase difference of the voltages impressed on the tubes, V_1 and V_2, accurately 90 deg. out of phase the input impedance of these tubes should be large. On this account it is best to use tubes with a low amplification constant.

At low frequencies the impedances of the stopping condensers, C_1 and C_2, may be comparab'e with the input impedances of the transformers, T_2 and T_3. These condensers should, therefore, have very nearly equal capacities. For the same reason the impedances of the coils, L_1 and L_2, should be large and have very nearly the same values.

As may be seen from Fig. 1, the resistance, R_2 is grounded at one point whereas R_1 is not. If the mutual capacity between the primary and secondary windings of the transformers, T_2 and T_3, is appreciable, then because of this dissymmetry, the current in R_1 will not be accurately 90 deg. out of phase with the current in R_2 even if the voltages impressed on the primaries of the transformers are 90 deg. out of phase. It is, therefore, preferable to place a grounded shield between the windings of these transformers. Inaccuracies on account of dissymmetry will not be eliminated altogether by this arrangement, but will be small, if the capacity between the shield and the secondary winding of the transformers is small and balanced.

The resistance, R_1 and R_2, should not be so large that the current through these resistances will be changed appreciably when shunted by the input impedance of the tube, V_3; for this reason this tube should also have a small amplification constant.

The telephone receiver, T, when in use, is necessarily partially grounded; hence, if there is any mutual capacity between the windings of the transformer, T_5,

current will flow from the primary winding through the secondary to ground and a false balance will be obtained. It is advantageous, therefore, to use here a transformer with a grounded shield between the windings. If the capacity of the primary winding to this shield is balanced, the current through the receiver, T, will in all cases be equal to zero whenever the drop of potential between K_3 and d is equal to V_x.

A Modification of the Circuit

Instead of a condenser, C, (Fig. 1) a mutual inductance may be used in the manner shown in Fig. 4. It is evident that, theoretically at least, the voltages impressed on the tubes, V_1 and V_2, will be 90 deg. out of phase as in the preceding arrangement. However, this circuit is less desirable because the electromotive forces impressed on the two tubes, will in practise not be 90 deg. out of phase on account of the distributed capacity in the inductance coils, which cannot be avoided entirely. A further disadvantage on the use

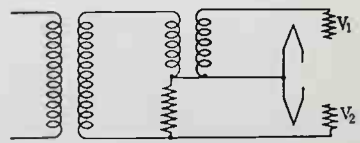

Fig. 4—Mutual Inductance Used in Place of Condenser

of inductance coils in place of a condenser is that stray magnetic fields may induce electromotive forces in other parts of the circuit and thus cause serious errors.

Mechanical Construction

Fig. 5 shows the general appearance of a form of potentiometer which was constructed on the principle just described. All the electrical parts of the apparatus are placed within a shielded box, to the vulcanite cover of which are attached the control dials and switches. For the various functions of these refer to Fig. 1. Dials 1, control the capacity of the condensers C; the maximum value of the capacity of this condenser is 1 microfarad. By means of the switch, 2, R may be given a value of either 1000 or 6000 ohms, the latter being used for frequencies below 200 cycles per second. Coarse and fine adjustments of the phase-shifting rheostat are made by the dials, 3. The reversing switches, 4, correspond to S_1 and S_2 of Fig. 1.

The ground glass window, 5, on which the light reflected from the mirror of the differential galvanometer is brought to a focus, serves to indicate when the voltages across the two coils of the galvanometer are equal in magnitude. A resistance dial, 10, controls the filament currents of the vacuum tubes. Dials, 6, regulate the slide wire resistance and so the potential drop between k_3 and d; the readings of these dials, after a balance has been obtained, thus give the magnitude of the measured e. m f. To measure the current flowing through the slide wire resistance an a-c. ammeter is connected to the terminals, 7. The measured voltage is brought to the potentiometer through the terminals, 8, and the telephone receiver or

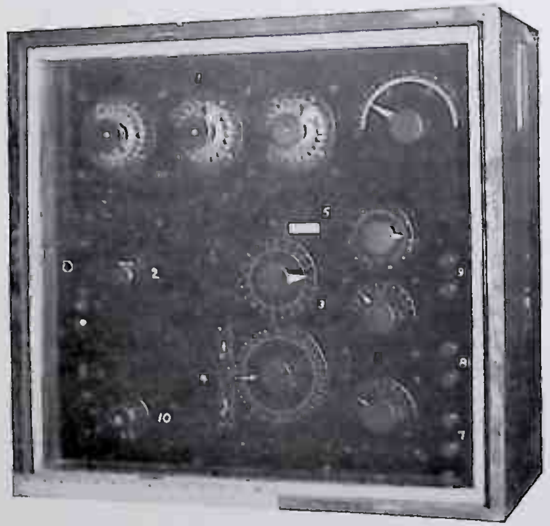

Fig. 5—Potentiometer

vibration galvanometer used in getting a balance is connected at 9.

Applications of the A-C. Potentiometer

The a-c. potentiometer possesses a number of distinctive advantages over other types of instruments for measuring alternating electromotive forces. One of the most important of these is the fact that its scale is linear, while the deflection of most a-c. measuring instruments, which depend for their action on a heating element or on electrodynamic forces, is proportional to the square of the electromotive force. Thus, since the sensitivity is the same over its whole scale, the potentiometer is particularly well adopted for measuring small electromotive forces on open circuit and in addition it can readily be constructed so as to give precise readings over a wide range of voltages. Electromotive forces ranging from 0.001 to 5 volts may be measured accurately with the potentiometer here described.

The a-c. potentiometer gives the voltage corresponding to the fundamental component only, regardless of the wave form of the electromotive force to be measured. This fact is a distinct advantage in certain classes of measurements, especially where values are to be obtained as a function of the frequency, as in many telephone problems, for whatever the wave form of the e. m. f. may be, effects due to the higher harmonics are eliminated.

In many alternating-current problems it is of fundamental importance to be able to measure phase. For example, in the study of distortion produced by amplifiers or other circuits in the transmission of currents of complex wave form, it is often necessary to determine the phase relations as well as the comparative magnitudes of the output and input currents at various frequencies. With the a-c. potentiometer here described both quantities are read directly after three simple adjustments.

The instrument is particularly well adapted for determining electromotive forces as a function of frequency, for, the differential galvanometer permits adjustment to be made quickly in going from one frequency to another. Measurements have been made, with this instrument and satisfactory results obtained for frequencies varying from 50 to 14,000 cycles per second.

The compensating potential difference is supplied by means of a slide wire resistance. This arrangement reduces errors on account of capacity to ground to a minimum. Only iron-core transformers and inductances placed within iron cases, are used in the construction of this potentiometer. Stray fields are thus eliminated, which in measurements at acoustic frequencies might have considerable effect on the precision of the apparatus. All the other parts may be enclosed completely in a shielded box. The apparatus is thus entirely protected from outside electrical disturbances.

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JUNE 1922

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ELECTROSTATIC TRANSMITTER

BY

E. C. WENTE

A DESCRIPTION OF A
WESTERN ELECTRIC ELECTROSTATIC TRANSMITTER
FOR MEASURING SOUND INTENSITIES
WITH DATA ON ITS SENSITIVITY AND PRECISION

REPRINT OF PAPER
Published in
THE PHYSICAL REVIEW
N. S., VOL. XIX., No. 5 MAY, 1922

THE SENSITIVITY AND PRECISION OF THE ELECTROSTATIC TRANSMITTER FOR MEASURING SOUND INTENSITIES.

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THE SENSITIVITY AND PRECISION OF THE ELECTROSTATIC TRANSMITTER FOR MEASURING SOUND INTENSITIES.

By E. C. Wente.

Synopsis.

Electrostatic Transmitter of Constant Sensitivity.—(1) *Characteristics.* This instrument is the same in principle as that described in 1917, but certain changes have been made which, as proved by actual tests, render the sensitivity independent of changes of temperature, pressure and humidity. The sensitivity is also found to be constant over a long period of time. By means of a piston-phone and a thermophone, for which corrected formulæ are available, both the absolute sensitivity and phase lag were determined for frequencies of from 10 to 12,000 cycles. Eight transmitters similarly constructed give the same curves within 20 per cent. With a steel diaphragm 0.0051 cm. thick having a natural frequency of 7,000 cycles, and with an air gap of 0.0025 cm., the mean sensitivity is about 0.35 millivolt/dyne. (2) *Use with an amplifier for measurement of sound intensity.* Combined with an amplifier of ordinary design the instrument has an over-all sensitivity which is practically uniform from 25 to 8,000 cycles. It is therefore remarkably well adapted for the measurement of the intensities of complex tones and tones of changing pitch and for use with an oscillograph for recording sound waves. On the other hand, if sounds of a definite pitch are to be measured, the apparatus can be made highly selective and almost any desired sensitivity can be obtained by using a tuned amplifier in connection with a vibration galvanometer.

The writer some years ago published a paper[1] in this journal on the use of an electrostatic transmitter for the absolute measurement of sound intensities. The transmitter, as there described, consisted essentially of a thin metal diaphragm under tension separated by a small distance from a plane metal plate, the plate and the diaphragm forming the two electrodes of an air condenser. Data were given which showed that the instrument had a uniform sensitivity over a wide range of frequencies and so was especially adapted for the measurement of intensities of sounds of various frequencies.

The relation between sensitivity and frequency was determined by means of a thermophone and also, at the lower frequencies, by means of a piston-phone. But, as the two methods did not yield the same absolute values, doubts were cast upon the accuracy of the results. Since then the theory of the thermophone has been further developed and corrected formulæ for its acoustic efficiency have been obtained.[2] The pressures calculated by these new formulæ have been found to agree with those

[1] Vol. X., No. 1, p. 39, July, 1917.

[2] E. C. Wente, Phys. Rev. (2), XIX, p. 666.

given by a piston-phone, so that an absolute calibration of a transmitter may now be made with the thermophone which is dependable over a wide frequency range.

The particular electrostatic transmitter described previously had a natural frequency of nearly 17,000 cycles per second. As this frequency is considerably beyond the region which is covered in most acoustic measurements, it is desirable to construct an instrument with a lower natural frequency and thereby obtain greater sensitivity. Dr. I. B. Crandall[1] has shown theoretically how the natural frequency and damping of an electrostatic transmitter may be given almost any desired values by cutting grooves of the proper size in the back plate.

The transmitter which is here described has a natural frequency slightly over 10,000 cycles per second and a damping constant of 14,000. The sensitivity-frequency characteristic is such that an amplifier may readily be constructed so that the sensitivity of the amplifier and transmitter combined shall be nearly the same from 25 to 8,000 cycles per second. Tests have shown that the sensitivity of the transmitter is closely maintained under various atmospheric conditions for a long period of time.

Construction of the Transmitter.

A sectional drawing of the transmitter is shown in Fig. 1. The transmitter differs from the instrument previously described in several essential respects. The diaphragm, *A*, is made of 0.002 inch (0.0051 cm.) steel and is stretched so that its natural frequency in free air is 7,000 cycles per second. Annular grooves are cut into the face of the back-plate, *B*, to give the diaphragm the desired natural frequency and damping. The length of the air-gap is 0.001 inch (0.0025 cm.). To keep out moisture, the space surrounding the back-plate is sealed off completely from the outside air. A thin rubber diaphragm, *C*, is provided to keep the pressure on the two sides of the steel diaphragm substantially equal under all conditions of temperature and atmospheric pressure.

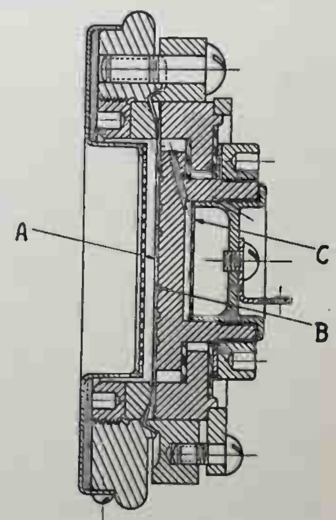

Fig. 1.

Sectional view of the electrostatic transmitter.

[1] E. C. Wente, Phys. Rev., (2) XI., 450 (1918).

Calibration of the Transmitter.

The open-circuit voltage of the transmitter per unit of pressure has been measured with the piston-phone for the frequency range of 10 to 200 cycles per second and with a thermophone for the frequency range of 60 to 12,000 cycles per second. Both of these instruments have been described in another paper.[1] The method of measurement was virtually the same as previously described,[2] except that an a.c. potentiometer[3] was used in place of a thermo-couple and galvanometer. The polarizing voltage in all cases was 200.

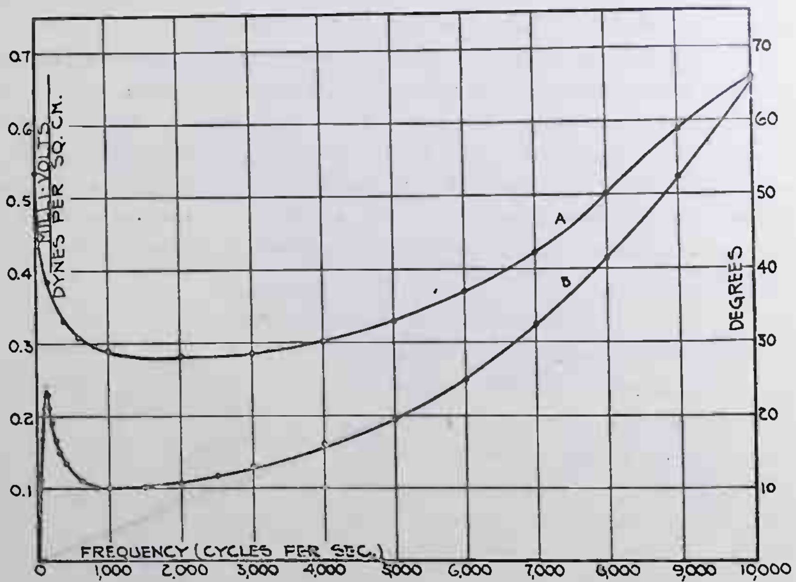

Fig. 2.

Sensitivity-frequency characteristic of the electrostatic transmitter. *A*. Volts per unit of pressure. *B*. Phase lag of e.m.f. behind pressure.

Fig. 2 gives the calibration curve obtained by averaging the results for eight transmitters. Between 25 and 8,000 cycles the calibration curve of none of the individual transmitters differs from this average curve by more than 20 per cent. To make all the transmitters exactly alike would require extreme precautions in construction, which would add considerably to their cost. However, even if there is a considerable variation in the mechanical constants of the individual instruments, we

[1] E. C. Wente, The Thermophone, loc. cit.

[2] E. C. Wente, Phys. Rev., (2) X., 51 (1917).

[3] E. C. Wente, Journal A. I. E. E., XL, p. 900, 1921.

may obtain a very nearly correct value of the calibration curve of any particular instrument by measuring the sensitivity at 200 cycles with the piston-phone and multiplying the ordinates of the curve given in Fig. 2 by the ratio of the value so obtained to the value of the ordinate of this curve at 200 cycles. For the eight calibrated transmitters the difference between the true calibration and the values obtained by this latter method was at no point greater than 12 per cent. for frequencies lying between 20 and 8,000 cycles per second, and for most of the transmitters the difference was nowhere greater than 5 per cent. This difference is small enough to be neglected in practically all acoustic measurements. Unless a precision greater than five per cent. is required it is therefore unnecessary to make any other measurements on transmitters of this type than that of the determination of the sensitivity at 200 cycles by means of the piston-phone, a measurement which may be made in a few minutes.

Fig. 2 shows that the sensitivity of the transmitter is not independent of frequency. However, since the transmitter is generally used with an amplifier, the sensitivity varies with the frequency in a desirable way. An amplifier as normally constructed has an amplification characteristic which is nearly proportional to the reciprocal of the efficiency characteristic of the transmitter. At any rate, an amplifier can readily be designed so that the sensitivity of the transmitter and amplifier combined is practically uniform from 25 to 8,000 cycles.

For the frequency range of 60 to 10,000 cycles a determination was also made of the phase relation between the pressure exerted on the diaphragm and the voltages generated by the transmitter. The values that were obtained are also plotted in Fig. 2. The maximum in the curve at 200 cycles is due to the fact that the damping of the diaphragm by the air in the gap increases with decrease in frequency.[1]

Variation of Sensitivity with Time.

To determine the change in sensitivity of the transmitter with time under ordinary conditions of use it would be necessary to make measurements extending over a long period. The more practical method of subjecting the transmitter to a higher temperature for a shorter period of time was therefore adopted. A transmitter was heated daily for five or six hours to about 45° C. and then allowed to cool to room temperature. No change in efficiency was observable even after this process had been continued for more than two weeks. The precision of the measurements was about 2 per cent. The transmitter was then heated

[1] I. B. Crandall, loc. cit.

to 100° C. for several hours. After being allowed to cool it was again tested but no change in sensitivity was observable. It is thus reasonable to assume that under ordinary conditions the sensitivity will not change appreciably during the course of several years.

Effect of Temperature and Pressure on Sensitivity.

When a rigid plate was used in place of the flexible rubber diaphragm, C, and equalization of the static pressure on the two sides of the diaphragm thereby prevented, measurements with the piston-phone and electrostatic voltmeter showed that the sensitivity of the transmitter changed about 2 per cent. per degree Centigrade at 200 cycles. With the rubber diaphragm, however, within the temperature range of 20 to 40 degrees C. the total change in sensitivity was less than 2 per cent., which is negligible for all practical purposes. The principal reason for this small temperature coefficient lies in the fact that the instrument is constructed almost entirely of the same material. The clamping rings and the diaphragm have nearly the same temperature coefficient of expansion, so that any change in temperature will produce but little change in the tension of the diaphragm. The diaphragm, although thin, cannot assume a temperature very different from that of the frame, for it is separated by such a small distance from the back plate that heat can flow from one to the other nearly as readily as if they were in contact.

If there were no displacement of the rubber diaphragm a change in temperature of 20° C. would produce a difference in pressure on the two sides of the diaphragm equal to 7 per cent. of one atmosphere. It follows, therefore, that a change in atmospheric pressure as great as this will not change the sensitivity of the transmitter by an appreciable amount.

Comparative Sensitivity of the Transmitter.

The mean value of the sensitivity of the transmitter given above is approximately 0.35 millivolt per dyne. A more comprehensible idea of this sensitivity may be obtained from the fact that the male voice in ordinary conversation exerts a pressure of about 10 dynes per sq. cm. at a distance of 3 cm. from the mouth of the speaker.[1] With a three-stage amplifier, thermocouple and galvanometer pressures as low as 0.01 dyne per sq. cm. may be measured. If the tone is produced by some electrical device such as a telephone receiver and the source of current is supplied by an oscillator, an a.c. potentiometer may be used. In this case pressures may be measured which are barely perceptible to the ear.

[1] I. B. Crandall and D. Mac Kenzie, Phys. Rev. (2), XIX, p. 221.

Concluding Remarks.

Little has been said about amplifiers in this paper although the electrostatic transmitter has little practical value unless it is used with an amplifier. Experiments have shown that the transmitter changes but little with time and atmospheric conditions. The amplifier, when properly designed and if the vacuum tubes are carefully chosen, will maintain the same value of amplification for a long period, provided the plate voltage and the filament current are kept at a constant value. An amplifier used in most of the experiments described above did not vary by more than a few per cent. during the course of several months. A combination of electrostatic transmitter and amplifier can thus be used as an absolute phonometer, the readings of which are dependable from day to day.

In the preceding discussion it has already been pointed out that one of the chief merits of the electrostatic transmitter is the fact that it has a very uniform response and so is especially adapted for measuring sounds of complex wave form or for comparing the intensities of tones of different frequencies. However, in some classes of problems it is desirable to have an instrument which is sharply tuned so that it will respond to a tone of one frequency and be unaffected by any other tones that may be present at the same time. If an electrostatic transmitter is connected to a tuned amplifier the output of which goes to a vibration galvanometer, selectivity of a very high order may be obtained and an amplifier of sufficient amplification may be used to give the combination almost any desired sensitivity.

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A CONDENSER TRANSMITTER AS A UNIFORMLY SENSITIVE INSTRUMENT FOR THE ABSOLUTE MEASUREMENT OF SOUND INTENSITY.

By E. C. Wente.

[Reprinted from the Physical Review, N. S., Vol. X. No. 1 July 1917.]

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By E. C. Wente.

THE various methods that have been used with more or less success for measuring the intensity of sound may be divided into five general classes: observation of the variation in index of refraction of the air by an optical interference method; measurement of the static pressure exerted on a reflecting wall; the use of a Rayleigh disc with a resonator; methods in which the motion of a diaphragm is observed by an optical method; the use of some type of telephone transmitter in connection with auxiliary electrical apparatus. The apparatus of either of the first two methods is non-resonant and hence the sensitiveness is fairly uniform over a wide range of frequencies. These methods are not sufficiently sensitive, however, to be of use in general acoustic measurements. On the other hand, instruments of the last three classes possess a natural frequency and are consequently very efficient in the resonance region. However, in the neighborhood of the resonant frequency the efficiency varies greatly with the pitch of the tone. It is possible to use a Rayleigh disc without a resonator, but its sensitiveness in that case is so low that it is of little practical value.

Because of the recent advances in the development of distortionless current amplifiers, the last class, in which use is made of some form of telephone transmitter, seems to offer the greatest possibilities. In the following pages a transmitter is described which has been calibrated in absolute terms for frequencies from 0 up to 10,000 periods per second and which has a nearly uniform sensibility over this range. The apparatus is easily portable, and possesses no delicate parts, so that, when once adjusted, it will remain so for a long period of time.

Except in cases where measurements are made with a single, continuous tone, it is desirable that the instrument for measuring the intensity of sound should have approximately the same sensibility over the entire range of frequencies used. This is especially important if the sound under investigation has a complex wave form. To avoid any great variation with frequency in the sensibility of a phonometer employing a vibrating system, it is necessary that the natural frequency lie outside the range of frequencies of the tones to be measured. Even if the natural frequency be compensated for in other ways, small variations in the constants of the instruments, which are always likely to occur, may change conditions appreciably at this frequency. It is pretty well recognized that for several reasons the natural frequency should lie above rather than below the acoustic range. If the instrument is to be used in studying speech, the natural frequency must indeed be very high. The upper limit of the frequencies occurring in speech is not definitely known, but it probably does not come below 8,000 periods a second. Titchener[1] found that if a Galton whistle was set so as to give a frequency of 8,500, the tone emitted could not be distinguished from an ordinary hiss.

An instrument that is to be used in studying speech should have high damping as well as a high natural frequency in order to reduce distortion due to transients. This is not so important if the natural frequency lies beyond the acoustic range, but nevertheless is desirable even in this case. Aperiodic damping is the best condition, but it is in general hard to obtain when the natural frequency is very high.

It seems best in this paper to give a rather complete treatment of the condenser instrument; for the sake of clearness, however, breaking up the matter into a number of sections as follows:

1. Theory of the Operation of an Electrostatic Transmitter.
2. General Features of the Design of the Instrument.
3. Deflection of the Diaphragm under a Static Force. Measurement of Tension and Airgap.
4. Sensitiveness of the Transmitter at Low Frequencies.
5. Sensitiveness at Higher Frequencies Determined by the Use of a Thermophone.
6. Natural Frequency and Damping of the Diaphragm.
7. Possibilities of Tuning.
8. Characteristic Features of the Instrument.
9. The Electrostatic Instrument used as a Standard Source of Sound.
10. Summary.

Some of these sections deal with theory and some with experimental work as need arises, the general aim being to put in proper order the material necessary for a full account of the condenser instrument.

1. Theory of the Operation of an Electro-static Transmitter.

The device to be described is a condenser transmitter, the capacity of which follows very closely the pressure variations in the sound waves. The use of such a device as a transmitter is not a new idea; in fact it

[1] Proc. Am. Phil. Soc., 53, p. 323.

was suggested almost as early as that of the corresponding electromagnetic instrument.[1] However, before good current amplifiers were available little or no use was made of electrostatic transmitters because of their comparatively low efficiency.

A simple circuit that may be used with such a transmitter is shown in Fig. 1. When the capacity of the transmitter is varied, there will be a corresponding drop of potential across R, which may be measured with an A.C. voltmeter or some other suitable device.

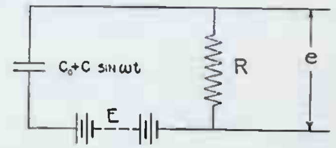

Fig. 1.

In order to get a quantitative expression for the magnitude of this voltage let us assume that the capacity at any instant is given by

$$C = C_0 + C_1 \sin \omega t,$$

in which $\omega = 2\pi \times$ frequency. For the circuit shown in Fig. 1

$$E - Ri = \frac{1}{C}\int i dt. \tag{1}$$

By differentiation and substitution we obtain

$$(C_0 + C_1 \sin \omega t)R\frac{di}{dt} + (1 + RC_1\omega \cos \omega t)i - EC_1\omega \cos \omega t = 0. \tag{2}$$

In order to evaluate this equation let us assume as a solution

$$i = \Sigma I_n \sin (n\omega t + \phi_n).$$

Substituting this value of i in (2) and determining the coefficients, we have

$$\left.\begin{aligned} i = {} & \frac{EC_1}{C_0\sqrt{\left(\frac{1}{C_0\omega}\right)^2 + R^2}}(\sin \omega t + \varphi_1) \\ & - \frac{EC_1^2R}{C_0^2\sqrt{\left[\left(\frac{1}{C_0\omega}\right)^2 + 4R^2\right]\left[\left(\frac{1}{C_0\omega}\right)^2 + R^2\right]}}\sin (2\omega t + \varphi_1 - \varphi_2) \\ & + \text{terms of higher order in } C_1/C_0, \end{aligned}\right\} \tag{3}$$

in which

$$\phi_1 = \tan^{-1}\frac{1}{C_0\omega R} \quad \text{and} \quad \varphi_2 = \tan^{-1}\frac{1}{2C_0\omega R}, \quad \text{etc.}$$

[1] La Lumiere Electrique, Vol. 3, p. 286, 1881.

For the best efficiency R should be made large in comparison with $1/C_0\omega$. In this case, the expression for the voltage e becomes

$$e = Ri = \frac{EC_1}{C_0} \sin (\omega t + \varphi_1) - \frac{EC_1^2}{2C_0^2} \sin (2\omega t + \varphi_1 - \varphi_2) + \cdots.$$

From this equation we see that in order to get a voltage of pure sine wave form for a harmonic variation of capacity, C_1 must be small in comparison with $2C_0$. This condition is satisfied as long as the A.C. voltage is small compared with E.

Retaining only the first term in (3) we have

$$e = Ri = \frac{EC_1R}{C_0\sqrt{\frac{1}{C_0^2\omega^2} + R^2}} \sin (\omega t + \varphi_1). \tag{4}$$

This equation shows that, so far as its operation in the circuit is concerned, the transmitter may be considered an alternating current generator giving an open circuit voltage $E(C_1/C_0) \sin (\omega t + \varphi_1)$ and having an internal impedance $1/C_0\omega$. It can also be shown that the transmitter can be regarded from this point of view if R is replaced by a leaky condenser or an inductance, so that this result may be said to be true in general.

2. General Features in the Design of the Instrument.

The general construction of the transmitter is shown in Fig. 2, from which the principal features are evident. The diaphragm is made of steel, 0.007 cm. in thickness, and is stretched nearly to its elastic limit. The condenser is formed by the plate B and the diaphragm. Since the diaphragm motion is greatest near the center, the voltage generated, which is proportional to C_1/C_0, will be greatest if the plate is small. On the other hand, since C_0 is proportional to the size of the plate, it cannot be made too small or the internal impedance of the transmitter will be too great. Therefore from the standpoint of efficiency, a compromise has to be made in determining the area of the plate. However, if it is made much smaller than the diaphragm, the natural frequency of the vibrating system will be decreased, as is explained below. On the basis of these factors the size of the plate indicated was judged to be about the best for the transmitter.

After some experiments with various dielectrics between the plate and the diaphragm it was concluded that air was most suitable. The dielectric constant of air is not so high as that of some other materials, but its insulating properties are better. However, the principal advantage of using air is, that it has a high minimum value of sparking

potential which lies in the neighborhood of 400 volts, below which there is no appreciable conduction. When E is less than this voltage, the air gap may be decreased without decreasing E, so that the efficiency of the instrument is limited practically only by the fact that when the gap is decreased below a certain value, the electrostatic force between the plate and diaphragm deflects the latter sufficiently to short circuit the condenser. When a potential difference of 320 volts was applied to the transmitter shown in Fig. 4, no appreciable current flowed across the air gap, certainly not more than 10^{-8} amperes. The fact that the air has such a high minimum sparking potential is one of the principal reasons why it is possible to design a successful condenser transmitter of the type shown in Fig. 2.

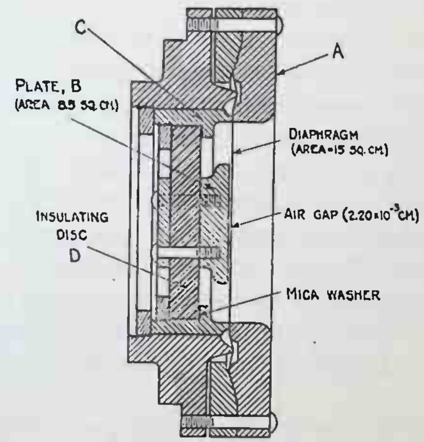

Fig. 2.

Sectional drawing of transmitter.

A word may be said in regard to the method of adjusting the transmitter so as to obtain a small uniform air gap. The surface of part A, next to the diaphragm, was ground plane before assembling. Small irregularities in the surface of the diaphragm facing the plate were removed by grinding with fine carborundum. Parts B, C and D were first assembled without the mica washer. The face of the plate and the ends of part C were then ground to the same level. Finally the mica washer was inserted between C and D and the whole apparatus assembled as shown. The mica may be split into washers of very even thickness, and a uniform air gap so obtained. The diaphragm is clamped between parts A and C, and is thus held in a true plane. In assembling the parts, the greatest care must be taken that no dust is caught between the plate and the diaphragm, for the insulation may be considerably reduced by the presence of any small particles in the gap.

Part C does not fit so perfectly against the diaphragm that the space surrounding the plate is shut off completely from the outside air. Changes in temperature and atmospheric pressure will therefore not affect the equilibrium position of the diaphragm.

The instrument used in these experiments was constructed just as shown in Fig. 2. It is evident from this figure that the diaphragm may be brought into contact with the plate if a mechanical pressure is accidentally exerted on the diaphragm. This will cause a spark to pass, if the transmitter was previously charged. In order to avoid damaging the metal surfaces in this way it may be advisable to glue to the face of the plate, A, a very thin layer of mica of uniform thickness, while still retaining an air-gap sufficient to allow free motion of the diaphragm.

3. Deflection of the diaphragm under a Static Force; Measurement of Tension and Air Gap.

It is not difficult to calculate the sensitiveness of the transmitter for low frequencies from the dimensions of its various parts, provided the magnitude of the deflection of every point of the diaphragm produced by a given static force is known. Since the diaphragm is made of very thin material and the tension is high, we may expect the diaphragm to behave very much as an ideal membrane, at least for frequencies near zero. In order to determine how closely this condition is approximated the following experiment was carried out.

When a static potential is applied between the plate and the diaphragm, the latter is deflected by the electrostatic force. The deflection produced

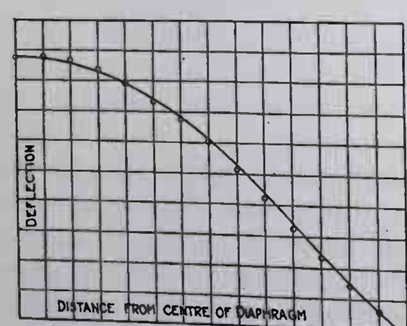

Fig. 3.

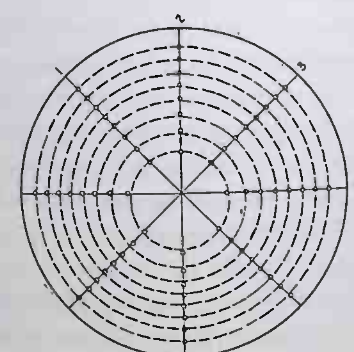

Fig. 4.

in this way by a known potential was measured by a device very similar to that used by Prof. D. C. Miller in his phonodeik.[1] By this arrangement the deflection of the diaphragm was magnified 30,000 times. The mean values of the deflections produced at various points along eight evenly spaced radii when a potential of 320 volts was applied are shown in Fig. 3. Points of equal displacement of the diaphragm are plotted in Fig. 4. The fact that the curves drawn through these points are

[1] D. C. Miller, Science of Musical Sounds, p. 79.

practically circles shows that the tension of the diaphragm was very nearly the same in all directions.

The distance between the plate and diaphragm was also measured with this apparatus by applying a mechanical force until the diaphragm touched the plate. The value obtained in this way was 2.20×10^{-3} cm. The capacity of the transmitter was measured on a capacity bridge and found to be 335×10^{-12} farads, from which the computed value of the air gap is 2.25×10^{-3} cm. The mean of the values obtained in these two ways is 2.22×10^{-3} cms.

In order to determine how closely the diaphragm approximates an ideal membrane, we may calculate the form that the latter would have assumed under the conditions of the preceding experiment.

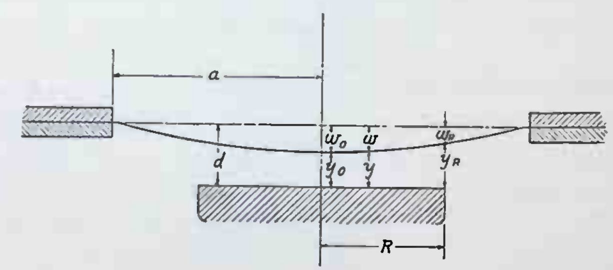

Fig. 5.

Referring to Fig. 5, if V is the potential between the plate and the diaphragm, and T, the tension of the membrane, we have

$$T\left[\frac{d^2w}{dr^2}+\frac{1}{r}\frac{dw}{dr}\right]+\frac{V^2}{8\pi y^2}=0.\text{[1]}$$

This relation holds from $r = 0$ to $r = R$. Let

$$A=\frac{V^2}{8\pi T}$$

and $x = \log r$, then since $(w - w_R) = (Y_R - y)$

$$\frac{d^2y}{dx^2}=A\frac{\epsilon^{2x}}{y},$$

or, since $(w - w_R)/(w_0 - w_R)$ is very nearly equal to $(R^2 - r^2)/r^2$ and $(w_0 - w_R)$ is small compared with y_0,

$$\frac{d^2y}{dx^2}=\frac{A}{y_0}\left(1-2k\frac{r^2}{R^2}\right)\epsilon^{2x},$$

in which

$$k=(w_0-w_R)/y_0.$$

[1] Rayleigh, Theory of Sound, II., p. 318.

From this we get

$$w = \frac{A}{4y_0^2}(R^2 - r^2)\left[1 - \frac{k}{2} - \frac{k}{2}\frac{r^2}{R^2}\right] + w_R. \tag{5}$$

The total force on the diaphragm is

$$F = \frac{\pi R^2 V^2}{8\pi d_1^2}$$

where d_1 is defined by the equation

$$\frac{\pi R^2}{d_1} = \int_0^R \frac{2\pi r dr}{y^2} = \frac{2\pi}{y_0}\int_0^R \left(1 - 2k\frac{r^2}{R^2}\right) r dr = \frac{\pi R^2}{y_0}(1 - k).$$

so that

$$F = \frac{\pi R^2 V^2}{8\pi y_0^2}(1 - k).$$

In the region extending from $r = R$ to $r = a$,

$$F = -2\pi r T\frac{dw}{dr}.$$

From this

$$w_R = \frac{AR^2(1 - k)}{2y_0}\log\frac{a}{R}. \tag{6}$$

From (5) and (6)

$$w = \frac{A}{4y_0^2}\left\{(R^2 - r^2)\left[1 - \frac{k}{2}\left(1 + \frac{r^2}{R^2}\right)\right] + 2R^2(1 - k)\log\frac{a}{R}\right\}. \tag{7}$$

This equation gives the form into which the diaphragm will be bent if it behaves like an ideal membrane. The curve representing this equation is shown in Fig. 3. The observed points do not lie very far from this curve. We therefore conclude that the diaphragm behaves sufficiently like an ideal membrane, so that no great error will be incurred if this assumption is made in calculating the sensitiveness of the transmitter for low frequencies.

From equation (7)

$$w_0 = \frac{V^2}{32\pi T y_0^2}\left\{R^2\left(1 - \frac{k}{2}\right) + 2R^2(1 - k)\log\frac{a}{R}\right\}$$

or

$$T = \frac{V^2 R^2}{32\pi y_0^2 w_0}\left\{\left(1 - \frac{k}{2}\right) + 2(1 - k)\log\frac{a}{R}\right\}. \tag{8}$$

Hence, if the deflection at the center of the diaphragm produced by a known voltage is measured, the tension may be calculated from (8). Results obtained in this way for the diaphragm used in these experiments are tabulated below.

Volts.	Deflection (w_0) (cm.).	Tension (T) (dynes)/(cm.).
200	6.0×10^{-5}	6.59×10^7
240	6.8	6.58
280	12.4	6.55
320	16.9	6.55
Mean.		6.57×10^7

4. SENSITIVENESS OF THE TRANSMITTER AT LOW FREQUENCIES.

Having satisfied ourselves that the diaphragm behaves sufficiently like a perfect membrane, and having determined the tension and air gap, we can now proceed to calculate the efficiency of the transmitter for low frequencies. To do this it is necessary to find the change in capacity produced by a given pressure on the diaphragm, since by equation (4) the voltage generated is proportional to C_1/C_0.

Referring to Fig. 5, we see that the capacity is $R^2/4d$ if the diaphragm is not deflected. From the curve of deformation when a potential is applied (Fig. 3), it is evident that w_R is very nearly equal to $0.45\ w_0$. Hence the air gap at any point is given by

$$d - w_0 + \frac{.55}{R^2}w_0 r^2,$$

and since the surface of the diaphragm deviates but little from a plane area, the normal capacity to the first approximation is

$$C_0 = \int_0^R \frac{2\pi r dr}{4\pi\left(d - w_0 + \frac{0.55 w_0}{R^2}r^2\right)} = \frac{R^2}{4d}\left[1 + \frac{w_0}{d} - \frac{.55 w_0}{2d}\right] = \frac{R^2}{4d'}, \tag{9}$$

in which d' may be called the effective air gap.

If a pressure, P, uniform all over the diaphragm produces a deflection, u, the capacity of the condenser will have been changed by the amount

$$C_1 = \int_0^R \frac{u \cdot 2\pi r dr}{4\pi y^2}. \tag{10}$$

The quantity in brackets of equation (9) does not differ greatly from unity in any practical case, so that no great error will be incurred if we set y in (10) equal to the constant value d'. Since

$$u = \frac{P}{4T}(a^2 - r^2),[1]$$

equation (16) may be written

$$C_1 = \frac{P}{8Td'^2}\int_0^R (a^2 - r^2) r dr = \frac{PR^2}{32Td'^2}[2a^2 - R^2]. \tag{11}$$

[1] Lamb, Dynamical Theory of Sound, p. 150.

C_1 is the change in capacity produced by a static pressure, P; but this differs very little from the maximum value of the alternating capacity resulting from a pressure, $P \sin \omega t$, provided $\omega/2\pi$ is small compared with the natural frequency of the diaphragm.

Having determined C_1 per unit value of P from equation (11), and C_0 from (9), we may calculate C_1/C_0 and hence the sensitiveness, *i. e.*, the volts per unit pressure. In practically all the experiments that have been made with the electrostatic transmitter, the D.C. voltage was 321. Under this condition we obtain 315 E.S.U. for C_0 from (9) and 1.96×10^{-3} E.S.U. per dyne per sq. cm. for C_1/P from (11). Hence we have for the sensitiveness

$$\frac{EC_1}{PC_0} = \frac{1.96 \times 10^{-3} \times 321}{315} = \frac{2.00 \times 10^{-3} \text{ volts}}{\text{dynes per sq. cm.}}.$$

In order to check this value directly by experiment, the apparatus diagrammatically shown in Fig. 6 was constructed. A receptacle was

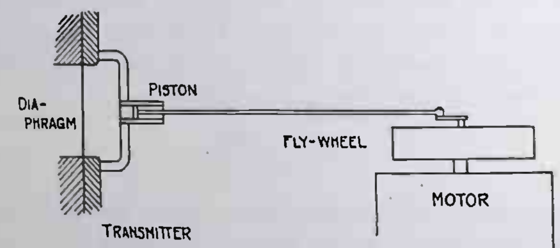

Fig. 6.

placed over the diaphragm as shown in the figure, thus forming an air-tight enclosure. Connected to this was a cylinder containing a piston. The connecting rod was long compared with the stroke of the piston so that with the motor running, the piston was given practically a simple harmonic motion. The fly wheel was fairly heavy and the connecting rod was made of stiff tubing, so that but little vibration was noticeable even when the motor ran at the highest speed.

The pressure variation is given by

$$\delta P = 1.4 P \frac{\delta V}{V},$$

in which δV is one half the total piston displacement and P is the maximum value of the alternating pressure.

$$V = 45.2 \text{ c.c. (volume of chamber)}$$

$$\delta V = \frac{0.68 \times 0.418}{2} = .142 \text{ c.c.}$$

Hence

$$\delta P = 1.4 \times 10^6 \times \frac{1.42}{45.2} = 4{,}400 \text{ dynes per cm.}^2$$

The root mean square value of the pressure is

$$\frac{4{,}400}{\sqrt{2}} = 3{,}120 \text{ dynes per cm.}^2.$$

The circuit used in this test is shown in Fig. 7. The electrostatic voltmeter had a very small capacity, giving it at low frequency an impedance large compared with the 80 megohm resistance in shunt.

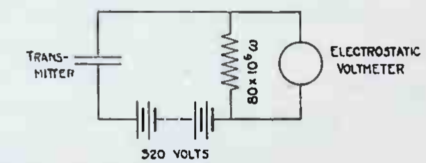

Fig. 7.

We may then calculate the open circuit voltage given by the transmitter from the voltmeter reading and the constants of the circuit, remembering that the transmitter may be regarded as a generator having an internal impedance $1/C_0\omega$. The following values were obtained in this way.

Motor Speed, R.P.M.	Frequency (P.P.S.).	Voltmeter Reading (Volts).	Open Circuit Volts.
1239	20.7	5.31	6.22
1074	17.9	5.31	6.27
950	15.8	5.31	6.33
824	13.75	5.20	6.29
584	9.75	4.92	6.27
Mean..........	....	....	........6.28

We therefore have for the sensitiveness,

$$\frac{6.28}{3120} = 2.02 \times 10^{-3} \frac{\text{Volts}}{\text{dynes per sq. cm.}}.$$

This value is in very close agreement with that given before, so that we may consider 2.00×10^{-3} volts per dyne as a reasonably correct value for the sensitiveness at low frequencies.

5. Sensitiveness at Higher Frequencies as Determined by the Use of a Thermophone.

By the methods just described the values of sensitiveness may be determined for very low frequencies only. In order to measure the sensitiveness at higher frequencies and also to get an idea of the natural frequency and damping of the vibrating system, use was made of the principle involved in the action of the thermophone as described by Arnold and Crandall.[1] A block of lead about 1.5 inches thick was placed against the face of the transmitter so as to form a cylindrical enclosure in front of the diaphragm, 1¾ inches in diameter and ⅜ inch long. The general arrangement is shown in Fig. 8. All crevices were sealed up so that the only openings to the cavity were two capillary tubes several inches long and of about 0.01 cm. bore. Two strips of gold foil were mounted symmetrically inside of this enclosure, the ends being clamped between small brass blocks. The supports were arranged in such a way that a current could be passed through the two strips in series. The connection between them was in electrical connection with the diaphragm.

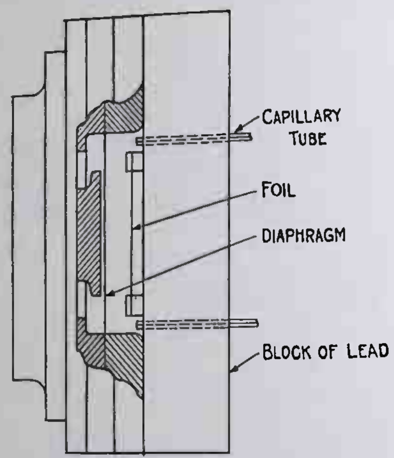

Fig. 8.

In the paper just cited it is shown that within an air-tight enclosure

$$\delta P = \frac{.0106 R i^2 P \sqrt{K_0}\left(\dfrac{\theta_1}{273}\right)^{1/4}}{\gamma V_0 \theta_2 f^{3/2}}, \tag{12}$$

in which

δP = maximum value of the alternating pressure within the enclosure.
P = normal pressure within the enclosure.
R = resistance of the foil.
i = r.m.s. value of the alternating current passing through the foil.
K_0 = diffusivity at 0° C. of the gas within the enclosure.
θ_1 = mean absolute temperature of the foil.
θ_2 = mean absolute temperature of the gas.
γ = heat capacity per unit area of the foil.
V_0 = volume of the enclosure.
f = frequency of the alternating current.

Equation (12) may be used for calculating the pressure variation provided the wave-length of sound is large compared with the dimensions of the enclosure. The velocity of sound in hydrogen is about four times

[1] Physical Review (Preceding Paper).

as great as in air; hence formula (12) holds for frequencies almost four times as high, when the enclosure is filled with hydrogen instead of air. Also, the diffusivity, K_0, is about six times as large for hydrogen as for air, so that greater pressure variation is obtained with the former. For these reasons, hydrogen was passed in a continuous stream through the enclosure by way of the capillary tubes, at a rate sufficiently slow to prevent any appreciable increase of the steady pressure above that of the atmosphere. The hydrogen was obtained from a Kipp generator and then passed through a solution of potassium permanganate and a drying tube containing phosphorus pentoxide.

In order to get the open circuit electromotive force of the transmitter, the circuit was arranged as in Fig. 9. The two resonant circuits were so

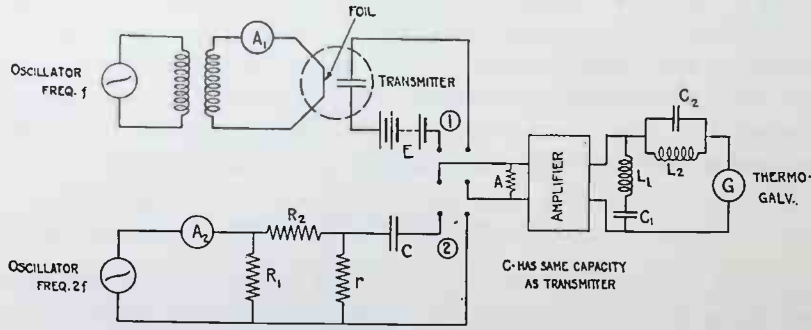

Fig. 9.

adjusted as to prevent current of the same frequency as that given by the oscillator from passing through the galvanometer. If a pure sine wave current passes through the foil, the pressure variation in the enclosure is of pure sine wave form and of double frequency. However, if there is any second harmonic present in the current, there will also be a component of the pressure variation of single frequency.[1] Putting in the resonant circuits eliminates this component from the measurements. The general procedure in making a measurement was as follows. The double-throw switch was first put in position 1, and the foil current and galvanometer current read. The switch was then thrown in position 2; R and r were then adjusted until the galvanometer read approximately the same as before. From the readings of A_2 and the values of R_1, R_2 and r, the voltage drop across r may be calculated. The open circuit voltage of the transmitter is then obtained by multiplying this voltage drop by the ratio of the galvanometer readings. That this gives us the open circuit voltage, follows from the fact that the transmitter

[1] Arnold & Crandall, *loc. cit.*

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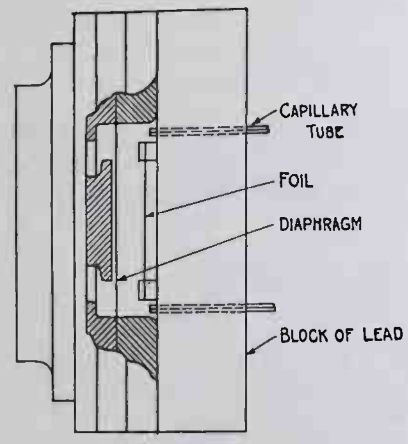

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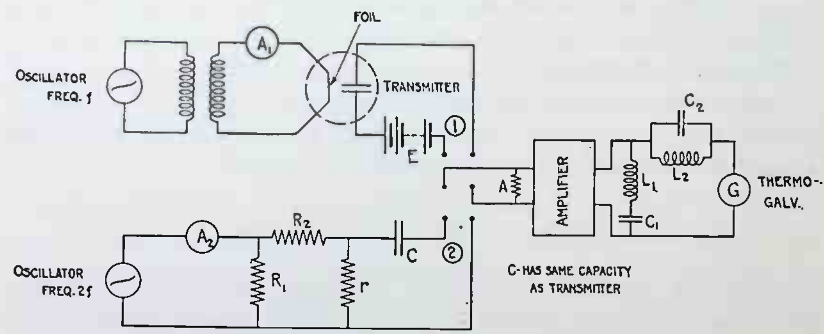

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[1] Arnold & Crandall, *loc. cit.*

behaves as a generator having an internal impedance $1/C_0\omega$. Oscillator No. 2, of course, is set at double the frequency of Oscillator No. 1.

The current passed through the gold foil was about 0.5 ampere at all frequencies. Resistance measurements showed that with this current density, the foil was not heated more than 10° C. above the room temperature. The values of the quantities entering into the formula (12) for this experiment were as follows:

$R = 4.18$ ohms.
$P = 10^6$ dynes/cm^2.
$K_0 = 1.48$ C.G.S. units.
$\theta_2 = 295°$.
$\theta_1 = 305°$.
$\gamma = 4.15 \times 10^{-6}$ calories per sq. cm.
Thickness of gold foil $= 7 \times 10^{-6}$ cm.
Width of each strip = 1 cm.
Length of each strip = 2.6 cm.

Substituting these values in equation (12), we have for the root mean square value of the alternating pressure

$$2.59 \times 10^6 \frac{i^2}{f^{3/2}} \text{ dynes per sq. cm.}$$

Dividing the measured open circuit voltage by this value should give us the volts per unit pressure for all frequencies within certain limits. Measurements were made in this manner for frequencies from 160 to 18,000 cycles per second. The general shape of the curve obtained by plotting these values is shown in Fig. 10.

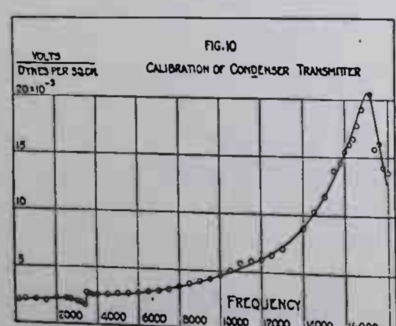

Fig. 10.

The absolute value of the sensitiveness at low frequencies as determined by this method was 0.121×10^{-3} volts per dyne, which is only about one sixteenth of that previously obtained by the piston method and by calculation from the dimensions of the instrument. In order to make further tests within the range of frequencies from 20 to 160 cycles, the gold was replaced by platinum foil, 4.42×10^{-4} cm. thick, and measurements were made as before. However, the size of the enclosure was increased in order to meet the conditions assumed in the derivation of formula (12), and for the same reason air was used instead of hydrogen.

Calculations made in a manner similar to that when gold foil was used gave a value of 1.93×10^{-3} volts per dyne per sq. cm. for the sensitiveness at low frequencies. This is in fair agreement with the value 2.00×10^{-3} obtained theoretically and with the piston apparatus.

Apparently when gold foil is immersed in hydrogen something takes place which is not taken account of in equation (12). The gold foil used was extremely thin (7×10^{-6} cm.) and when placed in hydrogen its specific heat per unit volume was apparently much greater than that of pure gold assumed in the calculations. On account of this discrepancy the gold leaf could not be relied upon for an absolute calibration, but it seemed reasonable to assume that the ratio between the true pressure and that calculated was independent of the frequency, so that a true relative calibration for different frequencies could be obtained. To get the absolute value of the efficiency at all frequencies, the values calculated from the readings on the gold foil were multiplied by the factor 2.0/.121 = 16.6. The results so obtained are those plotted in Fig. 10.

6. Natural Frequency and Damping of the Diaphragm.

It is thought that this curve (Fig. 10) may be relied upon to give the sensitiveness in absolute value for frequencies up to 10,000 cycles. Above this frequency the wave-length of sound approaches the diameter of the cylindrical enclosure. The wave-length in hydrogen at 10,000 cycles is 13 cm. whereas the greatest distance from boundary to boundary of the enclosure is 4.4 cm. Although the absolute values of the sensitiveness above 10,000 cycles are probably not given by the points plotted in Fig. 10, nevertheless, this curve indicates in a general way the behavior of the transmitter at high frequencies. The principal peak in this curve comes at 17,000 cycles, which undoubtedly corresponds to the natural frequency of the diaphragm. The damping cannot be determined with any great assurance of accuracy, although the curve as drawn would indicate a damping factor of the vibrating system of about six or seven thousand.[1]

These high values of natural frequency and damping are in a large measure due to the cushion effect of the air between the plate and the diaphragm. Free lateral motion of the air is prevented by its viscosity. This increases the rate of dissipation of energy when the diaphragm is vibrating and also adds to its elasticity.

To see whether 17,000 cycles is a reasonable value for the natural frequency we may make an approximate theoretical calculation. When

[1] The term damping factor as here used may be defined as the reciprocal of the time required for the amplitude to fall to 1/2.718 of its initial value.

the frequency is as high as 17,000 cycles it seems reasonable to assume that there is practically no lateral motion of the film of air. Let us further assume that the film of air is compressed and rarified adiabatically by the motion of the diaphragm and also that the plate is of the same size as the diaphragm. This latter condition is not quite satisfied in the case of the electro-static transmitter but no great error is introduced by this assumption, since the motion near the edge of the diaphragm is small. Under these conditions if d is the length of the air gap, P, the atmospheric pressure, ρ, the mass per unit area of the diaphragm, and T, the tension, the equation of motion of the diaphragm becomes:

$$\rho \frac{d^2w}{dt^2} = T\left(\frac{d^2w}{dr^2} + \frac{1}{r}\frac{dw}{dr}\right) - \frac{1.4Pw}{d},^{1}$$

or since w varies as $\epsilon^{j\omega t}$,

$$\frac{d^2w}{dr^2} + \frac{1}{r}\frac{dw}{dr} + \left(\frac{\rho\omega}{T} - \frac{1.4P}{Td}\right)w = 0. \tag{13}$$

The solution of (13), consistent with the boundary conditions, is

$$w = J_0(lr),$$

in which

$$l = \sqrt{\frac{\rho\omega^2}{T} - \frac{1.4P}{Td}}.$$

The boundary conditions require that $J_0(la) = 0$. The lowest root of this equation is 2.4 so that

$$\left[\rho\omega^2 - \frac{1.4P}{d}\right]\frac{a^2}{T} = (2.4)^2$$

or

$$f_0 = \frac{\omega}{2\pi} = \frac{1}{2\pi}\sqrt{\frac{T}{\rho}}\left(\frac{2.4}{a}\right)\sqrt{1 + \frac{1.4Pa^2}{Td(2.4)^2}}.$$

This equation gives the natural frequency of the diaphragm when vibrating in its fundamental mode.

For the transmitter used in the preceding tests —

$$T = 6.57 \times 10^7 \text{ dynes per cm.}$$

$$\rho = .05 \text{ gm. per sq. cm.}$$

$$a = 2.18 \text{ cm.}$$

$$P = 10^6 \text{ dynes per sq. cm.}$$

$$d = 2.22 \times 10^{-3} \text{ cm.}$$

Hence

$$f_0 = 6{,}350\sqrt{8.9} = 19{,}000 \text{ P.P.S.}$$

[1] Rayleigh, Theory of Sound, L., 318.

which is slightly higher than the observed value. With the plate removed, the diaphragm would have a natural frequency of 6,350. This shows that the film of air between the plate and the diaphragm increases the elastic factor many times. It is due entirely to this fact that it has been possible to obtain natural frequencies above 10,000 without making the diaphragm exceptionally small.

We may satisfy ourselves that the maximum point in the efficiency curve is not due to resonance in the cylindrical enclosure by calculating its resonant frequencies. These frequencies are determined by the equation

$$J_n'(\sqrt{K^2 - p^2\pi^2l^{-2}}\,R) = 0^{1}$$

in which

$$K = \frac{2}{a}\pi f_0.$$

a = velocity of sound.

l = length of cylinder.

R = radius of cylinder.

p = an integer.

Since the foil was placed symmetrically in the enclosure, only the symmetrical modes of vibration need be considered, in which case $n = 0$.

The first root of the equation $J_0'(Z) = 0$ is 3.83. For the lowest resonant frequency, $p = 0$, so that we have

$$f_0 = \frac{a}{2} \cdot \frac{3.83}{R}.$$

In this problem

a = 127,000 cm./sec. (velocity of sound in hydrogen).

R = 2.18 cm.

hence

$$f_0 = 35{,}500 \text{ cycles per second.}$$

which is very much above the frequencies covered in the calibration.

If the enclosure is filled with air instead of hydrogen, the first resonant frequency comes at about one fourth of 35,500 or 9,000 p.p.s. A series of measurements were made with the circuit arranged as in Fig. 9, and air instead of hydrogen surrounding the gold foil. Points were calculated and plotted; the curve so obtained showed a sharp resonant point at 9,600 but none below. This may be taken as further evidence that the maximum point in Fig. 10 is not due to any resonance in the enclosure and so corresponds to the natural frequency of the diaphragm.

[1] Rayleigh, Theory of Sound, II, 300.

There is an irregularity in the calibration at about 3,500 periods per second. This is undoubtedly due to the natural frequency of the back-piece. At any rate, vibration of the plate would have an effect of this general character, *i. e.*, the efficiency would be decreased below, and increased above, resonance. In a later design, the plate and support have been made more rigid so as to form practically one solid piece. It is believed that with the newer model, the irregularity in the curve will have been eliminated.

This completes the account of the experimental work done in calibrating the instrument.

In order to obtain some idea of the sensitiveness of the electrostatic transmitter just described as compared with an electromagnetic instrument, the sensitiveness of the former was compared directly with an ordinary telephone receiver used as a transmitter, over a considerable range of frequencies. Except within a hundred cycles of the resonant frequency of the diaphragm of the receiver the electrostatic transmitter was found to generate a greater voltage for a given sound intensity.

7. Possibilities of Tuning.

Since an electrostatic transmitter is equivalent to an alternating current generator having an internal impedance $1/C_0\omega$, it is evident that, if in the circuit shown in Fig. 1, the resistance R is replaced by an inductance L, the voltage e will be a maximum for a frequency of

$$f = \frac{1}{2\pi\sqrt{LC_0}}.$$

The sharpness of tuning will of course depend upon the possibility of getting an inductance with a small resistance. In many problems in acoustics it is desirable to have a tuned system and in that case it is also better to have a diaphragm of low natural frequency and damping.

In order to get an expression for the sensitiveness as a function of the frequency, let us assume that we have a parallel plate condenser, one of the plates of which is fixed and the other moved perpendicularly to its own plane by a simple harmonic force. Practically this condition is approximated by a diaphragm, the center of which is separated a short distance from a plane plate as is shown in Fig. 2.

Let x = displacement of the diaphragm from its equilibrium position.
d = air gap, assumed large compared with x.

Then

$$\frac{1}{C} = \frac{1}{C_0}\left(1 + \frac{x}{d}\right).$$

The mechanical impedance of the diaphragm is

$$Z_1 = r + j\left(m\omega - \frac{s}{\omega}\right),$$

where r = resistance factor,
m = mass factor,
s = elasticity factor,
$\omega = 2\pi \times$ frequency.

If T = kinetic energy of the entire system,
W = potential energy of the entire system,
$2F$ = rate of dissipation of energy,

then

$$2T = m\dot{x}^2 + L\dot{y}^2,$$

$$2W = sx^2 + \frac{1}{C_0}\left(1 + \frac{x}{d}\right)(Y + y)^2,$$

$$2F = R\dot{y}^2 + r\dot{x}^2,$$

where Y is the permanent, and y the variable electric charge on the condenser. The equations of motion for the system are

$$\frac{d}{dt}\left(\frac{\partial T}{\partial \dot{x}}\right) - \frac{\partial T}{\partial x} + \frac{\partial F}{\partial \dot{x}} + \frac{\partial W}{dx} = P\epsilon^{j\omega t},$$

$$\frac{d}{dt}\left(\frac{\partial T}{\partial \dot{y}}\right) - \frac{\partial T}{\partial y} + \frac{\partial F}{\partial \dot{y}} + \frac{\partial W}{\partial y} = 0.$$

If second order quantities are neglected, and also the constant terms, which affect only the equilibrium position, these equations become

$$\left.\begin{aligned} pm\dot{x} + r\dot{x} + \frac{s}{p}\dot{x} + \frac{Y\dot{y}}{pC_0d} &= P \\ pL\dot{y} + R\dot{y} + \frac{1}{pC_0}\dot{y} + \frac{Y}{pC_0}\frac{\dot{x}}{d} &= 0 \end{aligned}\right\} \qquad (14)$$

in which p is written for

$$j\omega = \frac{d}{dt},$$

solving equations (14) for $\dot{y}$ and substituting the values,

$$E = Y/C_0, \quad Z_1 = (r + s/p + pm), \quad Z = (R + 1/pC_0 + RL),$$

we have

$$\dot{y} = \frac{PE}{pd[(E/pd)^2 - Z_1Z]},$$

or

$$e = \dot{y}(R + pL) = \frac{PE(R + pL)}{pd[(E/pd)^2 - Z_1 Z]}.$$

In any practical case $(E/pd)^2$ is small compared with Z_1Z so that we may write without much error

$$e = \frac{EP(R + pL)}{pdZ_1Z}.$$

In order to obtain a large value of e/P, which is a measure of sensitivness, Z_1Z should be made small, *i. e.*, the diaphragm should have a natural frequency equal to the frequency, $\omega/2\pi$, and the electrical circuit should be in resonance at the same point.

No extensive measurements have been made with the circuit arranged in this way, although enough has been done to show that it is feasible in some cases. The chief difficulty lies in the fact that the transmitter capacity is so small that the inductance has to be very large to get resonance for ordinary sound frequencies. This difficulty may be overcome by shunting the transmitter with a condenser, which of course reduces the generated voltage.

8. Characteristic Features of the Instrument.

Because of the high internal impedance of the electrostatic transmitter, it is possible to use the instrument efficiently only with high impedance apparatus, such as an electrostatic voltmeter or a vacuum tube amplifier. However, this is no special disadvantage if an amplifier is to be used, because it is not desirable to use a transformer in connection with an instrument for measuring sound intensities, since the ratio of transformation of a transformer is not independent of either frequency or load.

The method for calibration of this instrument as explained in the preceding pages is rather elaborate and requires considerable care. But since the efficiency depends primarily on the air gap and tension, it should not be difficult to make duplicate transmitters to which the same calibration applies, since the desired values of air gap and tension may be obtained without great difficulty, the former being tested by measuring the capacity, and the latter by determining the deflection produced by a known potential between the plate and diaphragm.

The fact that the sensitiveness of this instrument is independent of any properties of material, such as magnetization or electrical resistance, is of considerable advantage. For this not only allows us to make instruments which are almost exact duplicates, and so let the calibration for

one instrument serve for all the rest, but, the calibration is also constant with the time. The metal parts are of machine steel throughout; from the construction as shown in Fig. 2, it is therefore evident that temperature can affect the sensitiveness but little. The tension of the diaphragm is, of course, not absolutely independent of temperature, nor is the action of the cushion of air between the plate and the diaphragm independent of the barometric pressure: but these effects are hardly worth considering. Being made of heavy material, the transmitter satisfies the requirement in the way of ruggedness; having once been adjusted, it should remain so, even if subjected to considerable rough usage.

The sensitiveness of the transmitter is not absolutely uniform, but varies only about a hundred per cent. between zero and 10,000 cycles, as the curve in Fig. 10 shows. This variation is much less than would be the case with an electromagnetic instrument with a diaphragm having the same natural frequency and damping. Except for eddy current and iron losses, the voltage generated by an electromagnetic transmitter is proportional to the *velocity* of the diaphragm, whereas that given by the electrostatic transmitter is proportional to the *amplitude*. Below the natural frequency, the variation of velocity with frequency is much greater than the variation of amplitude since the velocity is proportional to the product of the frequency and amplitude.

In most problems the transmitter would be used with an amplifier. Now, the sensitiveness of the transmitter increases, whereas the efficiency of an amplifier sometimes decreases with the frequency; at any rate, it is possible to design a circuit for the amplifier, so that the combination of the two has a constant sensitiveness over a wide range of frequencies.

Since the natural frequency of the transmitter is very high, instantaneous records of sound waves obtained in combination with a distortionless oscillograph would not only give the relative amplitudes of the different frequencies into which the sound may be analyzed, but also the phase relations should be practically unchanged for frequencies up to 10,000 p.p.s.

As yet no instrument is available which will record without distortion currents of frequencies as high as 10,000 cycles. Only after such an instrument has been developed will it be possible to get a true record of consonant sounds. The same is true in regard to the quantitative study of the quality of musical instruments. However, by using an ordinary high frequency oscillograph in connection with a condenser transmitter and amplifier, it should be possible to get curves equal to or better than any obtained heretofore.

9. THE ELECTROSTATIC INSTRUMENT USED AS A STANDARD SOURCE OF SOUND.

There is of course no theoretical reason why the instrument described in the preceding pages cannot be used in a reversible manner: that is, as a source of sound when an alternating voltage is applied between the plate and the diaphragm. If the instrument is to be used in this way, it is better to have the plate the same size as the diaphragm, in order to get the maximum electrostatic force for a given voltage and air gap. The resulting increase in capacity is in general no disadvantage in this case. Also for convenience in using the instrument it may be desirable to have the face of the plate covered with a thin layer of mica.

Because of the simplicity of this type of instrument it is not difficult to calculate the output of sound energy for a given voltage *after its efficiency as a transmitter has been determined.* It is evident that the instrument can be excited in two different ways; (*a*) the alternating voltage can be applied alone, and (*b*) it may be superimposed on a static potential maintained by a battery in exactly the same way as when the instrument is used as a transmitter. The main principles underlying the two kinds of excitation in this case are quite similar to those discussed by Arnold and Crandall in connection with the excitation of the thermophone by pure A.C. and by A.C. with D.C. superimposed. For this reason neither type of excitation need be discussed at length; but a brief treatment of the condenser instrument excited by pure alternating current will be given.

When a pure alternating voltage is applied, the mean deflection of the diaphragm will depend on the magnitude of this voltage and the efficiency may vary somewhat because of the change in mean air gap, and the consequent change in the cushion effect of the air sheet on the motion of the diaphragm. It is therefore necessary to have curves corresponding to the curve in Fig. 10 but for a series of applied static potentials. These are most easily obtained by determining for a number of frequencies the generated voltage as a function of the static potential when sound of a fixed intensity falls on the transmitter. It will be found that the alternating voltage generated is so nearly proportional to the static potential that for most acoustic work this may be assumed to be the case.

When an alternating potential $\sqrt{2}v \sin \omega t$ is applied to the plates, the electrostatic force per unit area acting on the diaphragm is

$$\frac{v^2}{8\pi d^2}(1 - \cos 2\omega t). \tag{15}$$

Now refer to Fig. 10, assuming that the curve there shown gives the efficiency of the instrument, *used as a transmitter*, for an applied static potential v. If we multiply the ordinate (*i. e.*, the voltage per unit pressure) at frequency $\omega/\pi = f$ by the quantity

$$\frac{C_0}{v} \equiv \frac{C_0}{E}$$

we can obtain (cf. (4)) $\overline{C}_f$, the change in capacity per unit pressure. The total change in capacity due to the electrostatic force is then, (if $\overline{C}$ is the change per unit pressure at zero frequency)

$$C_1 = \frac{v^2}{8\pi d^2}(\overline{C} - \overline{C}_f \cos 2\omega t) \tag{16}$$

from which we can proceed to calculate the amplitude of motion of the diaphragm.

It is necessary of course to have a mean value of d, the air gap, but it is sufficiently accurate to take an arithmetic mean of the values at the center and at the edge of the diaphragm. The motion at the center is greater, but the motion near the edges extends over a greater area.

In computing the mean amplitude of the diaphragm we shall introduce very little error if we take the form of the diaphragm as that of a paraboloid. u, the amplitude at any radius, r, is given by the relation already quoted

$$u = \frac{P}{4T}(a^2 - r^2), \tag{17}$$

in which a = the radius of the diaphragm and plate. Equation (11) gives the total change in capacity in terms of P and T, that is (since $a = R$)

$$C_1 = \frac{a^4}{32d^2}\frac{P}{T}, \tag{11'}$$

or, eliminating P/T between (11′) and (17) we have, for displacement at any radial distance r, in terms of maximum capacity change

$$u = \frac{8d^2}{a^4}(a^2 - r^2)C_1.$$

Substituting for C_1 the value given in (16) we have

$$u = \frac{v^2}{\pi a^4}(a^2 - r^2)(\overline{C} - \overline{C}_f \cos 2\omega t), \tag{18}$$

in which v is the r.m.s. value of the applied alternating voltage, and $\overline{C}_f$

is the change in capacity per unit pressure, determined in the manner described from the calibration curve of the instrument used as a transmitter. Equation (18) is rigorously true for all frequencies within the range of calibration, because the quantity $\bar{C}_f$ is taken from the calibration curve.

If, however, T is known, we can obtain an approximate value of u good at low frequencies, without any knowledge of C_1. (This is merely "equilibrium theory" and makes use only of the elastic factor, leaving the inertia and mechanical resistance of the moving system out of account). Substituting the value of electrostatic force (15) for P in (17) we have

$$u = \frac{v^2(a^2 - r^2)}{32\pi d^2 T}(1 - \cos 2\omega t). \tag{19}$$

The actual acoustic effect may be determined by the usual methods. If the diaphragm forms a wall of a small enclosure, the intensity is determined by the ratio of the volume displaced by the diaphragm as it vibrates to the volume of the enclosure. In other cases the intensity at a given point is calculated by determining the velocity potential due to the motion of the diaphragm.

It has been tacitly assumed that the amplitude of motion of the diaphragm is small compared with the air gap. This is necessary in order to get a pure tone when a sine wave E.M.F. is applied. While the instrument will not take care of a very large amount of energy, sound of the same order of intensity may be obtained as from an ordinary telephone receiver without appreciable distortion.

SUMMARY.

1. A description is given of a transmitter of the electrostatic type which is especially adapted for measurement of sound intensities over a wide range of frequencies. The instrument is portable and is sufficiently rugged to retain its calibration.

2. A discussion is given of the necessary auxiliary apparatus and the precautions necessary for proper use.

3. A theory of the transmitter has been developed by which its operation can be predicted from a few simple measurements.

4. A description is given of the calibration of such an instrument in absolute terms over a wide range of frequencies. It is found that its efficiency may be made practically uniform for frequencies up to 10,000 cycles per second, and the results of the calibration are in agreement with the theory.

5. The apparatus when once adjusted may be used for the measurement of the intensities of sound at any frequencies throughout this wide range without further special adjustment.

6. Due to the uniform response through this wide frequency range it will be possible to secure correct indications of complex wave forms and to determine not only the relative intensities of the components but also their phase differences.

7. When properly calibrated this apparatus can be used as a precision source of sound.

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